MOTION PICTURE SOUND ENGINEERING

Motion Picture Sound Engineering

A Series of Lectures
presented to the classes enrolled in the
courses in Sound Engineering given by the

RESEARCH COUNCIL
of the
ACADEMY OF MOTION PICTURE ARTS AND SCIENCES
HOLLYWOOD, CALIFORNIA

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To the memory

of

Irving G. Thalberg

who

inaugurated the Research Council

PREFACE

In the period which has elapsed since the publication of the book "Recording Sound for Motion Pictures," by the Academy of Motion Picture Arts and Sciences in the year 1931, there have been many advances in the development of equipment and technique for recording sound for the motion picture.

Most of these advances have taken place as a result of the cooperative endeavor of a large number of technical experts, and this book covering current sound recording and sound reproducing practice results from a further extension of the technical cooperative idea as exemplified by the Research Council of the Academy of Motion Picture Arts and Sciences.

Although the Academy of Motion Picture Arts and Sciences has engaged in cooperative research ever since its organization in 1927, the Research Council as it now functions was organized in 1934, having at that time eight cooperative projects in the hands of eight separate Committees. Founded upon the recognition of the value of a group judgment based upon an orderly study of all of the facts, the Council was set up to coordinate technical problems within the industry, the entire effort being directed towards getting pictures of better quality upon the screen and getting them there at the lowest net cost and with the highest net efficiency.

In the Council the motion picture industry has available a smoothly working machinery to handle industry projects involving investigation beyond the scope of any one individual studio or company and of tackling problems which can be dealt with more effectively and more economically through cooperative action rather than by the individual companies working separately and thus duplicating their development effort and expense.

Since its reorganization in 1934, the Council's activities have steadily grown until at the present time there are thirty-six technical committees operating under Council sponsorship, investigating problems of sound recording, sound reproduction, projection, laboratory practice, film preservation, photography, lighting and set acoustics.

The work of all of the various technical committees functions through the Council proper, which consists of a chairman representing

the producing companies and one technical representative for each or the studios participating in this work—these being: John Livadary, representing Columbia Pictures Corporation; Douglas Shearer, representing Metro-Goldwyn-Mayer Studios; Farciot Edouart, representing Paramount Productions; John Aalberg, RKO-Radio Pictures; E. H. Hansen, 20th Century-Fox Film Corporation; Thomas Moulton, United Artists Studios; Homer Tasker, Universal Pictures Corporation; and Nathan Levinson, Warner Brothers-First National Studios.

The Council acts as the governing body directing the work and activities of the Committees, each of which in turn operates under the direction of a chairman who is selected from the staff of one or another of the participating companies and chosen for his particular qualifications in the field within which the Committee is to function.

The members of the Council, and the chairmen and members of the various committees all donate their time and knowledge to carry on this cooperative activity, receiving no compensation other than the satisfaction of having participated in a worthwhile industry activity and of having achieved results which without their participation would have been impossible.

We believe that this book, which has resulted from approximately two years' work by the Council's Committee on Industrial Education and the group of sound course instructors and authors who have prepared materials included in it, will become a valuable addition to the technical literature on the motion picture and we offer it to the industry with a feeling of great pride.

Wm Koenig

Chairman,
Research Council
Academy of Motion Picture Arts and Sciences.

Hollywood, California.
January 3, 1938.

ACKNOWLEDGMENTS

In presenting MOTION PICTURE SOUND ENGINEERING to the industry the Research Council of the Academy of Motion Picture Arts and Sciences acknowledges the cooperation of A. P. Hill, of Electrical Research Products, Inc., who prepared the lectures given in the two courses in the Fundamentals of Sound Recording, one in the fall of 1936 and the other in the spring of 1937, and Messrs. Fred Albin, of United Artists Studios, L. E. Clark, of the RCA Manufacturing Co., and John Hilliard and Harry Kimball of Metro-Goldwyn-Mayer Studios, who prepared the lectures presented in the Advanced Course in Sound Recording given during the spring of 1937.

In addition, the Council and the industry are indebted to Electrical Research Products, Inc., the Metro-Goldwyn-Mayer Studios, the RCA Manufacturing Company, and the United Artists Studios for their helpful cooperation in making it possible for these experts to undertake the responsibility of these Courses and perform this service to the motion picture industry.

The time intervening since the close of the last classes of the courses and the issuance of this book in its final form has not been idle. Early in this period a policy of revision and expansion was decided upon, in order that MOTION PICTURE SOUND ENGINEERING might, in its final form, deal as completely and authoritatively as possible with the subjects with which it is concerned.

Several chapters not presented originally to the sound course classes were written for inclusion in the book by Wesley C. Miller and Kenneth Lambert in order that the text material might completely cover the subject of motion picture sound recording and reproducing.

Exhaustive editing and criticism of one another's work by each of the group of authors, helped to shape the book into its final form.

The Research Council's Committee on Industrial Education under the Chairmanship of Dr. J. G. Frayne, assisted by Barton Kreuzer, Dr. Burton F. Miller, William Thayer and Ralph Townsend has served to guide the instructors, both in their presentation of the lecture material to the various classes, and in the preparation of the book itself.

The Council can hardly make adequate acknowledgment to those few individuals who, in addition to the authors, have throughout the

period of writing and editing of the book given of their time, thought and energy to help in its preparation. To these few, particular thanks are due: Wesley C. Miller, Nathan Levinson, and John Hilliard.

In addition to the authors and others mentioned above, the Council is indebted to the following who have helped in one way or another in the preparation of this book: W. E. Beatty of Warner Brothers-First National Studios, Barton Kreuzer, Wallace V. Wolfe, and Michael Rettinger of the RCA Manufacturing Company, and K. F. Morgan, Don Loye, W. H. Garrison, and R. W. Wight of Electrical Research Products, Inc.

In addition, acknowledgment is made to the Acoustical Society of America, the American Institute of Electrical Engineers, the Society of Motion Picture Engineers, the Institute of Radio Engineers, the Bell System Technical Journal, the Brush Development Company, Electrical Research Products, Inc., the RCA Manufacturing Company, P. Blakiston's Son & Co., Inc., McGraw-Hill Book Company, Inc., the University of Chicago Press, John Wiley and Sons, and Dr. John F. Blackburn for their cooperation in making available illustrations used throughout this text.

In addition, the editing and proofreading of the book has benefited to a very considerable degree by the able and conscientious assistance of William F. Kelley of the Research Council staff.

Gordon S. Mitchell

Manager,
Research Council
Academy of Motion Picture Arts and Sciences

January 3, 1938.

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FOREWORD

Early in 1936 the Research Council of the Academy of Motion Picture Arts and Sciences, recognizing the great advancements in the sound field during the previous few years and realizing the value to the industry of a completely trained personnel in the studios, appointed the Council's Committee on Industrial Education, under the chairmanship of Dr. J. G. Frayne, assisted by Barton Kreuzer, Dr. Burton F. Miller, William Thayer and Ralph Townsend, to investigate the needs for vocational education within the motion picture studios.

This committee subsequently arranged and conducted a Course in the Fundamentals of Sound Recording, which was attended by 100 selected studio sound department employees, divided into two classes of 50 each.

Both of these classes, meeting twice each week, were instructed by A. P. Hill of Electrical Research Products, Inc.

Enrollments in this first course in the Fundamentals of Sound Recording were limited to 100, each studio's quota being chosen by the director of that studio's sound department. The first announcement of the course calling for applications for enrollment to be submitted through the studio sound directors, resulted in the filing of 206 applications. In consideration of this unforseen popularity of the Council's educational program, it was decided to repeat the Fundamental Course again the next fall, under the direction of Mr. Hill, to an additional group of 50 sound technicians.

Concurrent with the second of the Fundamental Courses, the Advanced Course in Sound Recording, under the direction of Messrs. Fred Albin, L. E. Clark, John Hilliard and Harry Kimball was arranged for a group of 50 studio sound engineers consisting of those who had completed the first Fundamental Course, and others having the necessary previous education and experience.

This educational program has proven to be of great value to the 200 studio sound department employees who were enrolled, each of whom is of greater value to the industry because of the added knowledge gained from the Courses.

Stenographic transcripts were made of the lectures presented in these Courses, and material included in this book has been assembled from these transcripts.

FOREWORD

Any discontinuity in the text results from the fact that the book is made up of a series of lectures which were originally prepared for presentation to the sound course classes and subsequently edited by their authors for inclusion in this work.

We believe that MOTION PICTURE SOUND ENGINEERING will answer a definite need for an authoritative treatise on sound recording and sound reproducing, and anticipate that this volume will become an important reference authority on sound in motion pictures.

Nathan Levinson

Vice-Chairman,
Research Council
Academy of Motion Picture Arts and Sciences.

Hollywood, California.
January 3, 1938.

In Appreciation

The individuals listed on the following pages, by serving on various Committees and by participating in its activities, have contributed greatly to the success of the Research Council of the Academy of Motion Picture Arts and Sciences, and through their constant interest and activity the Council is able to continue its record of achievement.

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PART I

Chapter I

BASIS OF MOTION PICTURE SOUND

By WESLEY C. MILLER

"The world is so full of a number of things, I'm sure we should all be as happy as kings." Even though the human race fails to attain to this condition of perfection each of us has his own ideas about the world in which he lives. Consciously and otherwise we are continually absorbing from the infinite variety which surrounds us, and in turn we make our contribution through our own reflected personality. Since earliest childhood we have been sorting out and compiling impressions and associations which are limited only by our own imagination, reasoning power and environment. As the conscious mind learns to exert its main effort in directing our activities, much of this assimilation becomes involuntary and automatic. The accumulation of impressions is important — indifference to the means of accumulation results. In normal life this indifference is not necessarily a criticism. However, in the creation of artistic illusions such as the motion picture, careful analysis is required of the nature of both the voluntary and involuntary sensory impressions.

Translation of things into impressions in the consciousness of our brains and minds is through the senses. The sensations which these organs furnish to us have enabled us to build up an accumulation of experience whereby cause and effect assume the relations and associations we have learned to expect. In the natural world certain combinations of objective elements of sight and sound are familiar to us. It is only when our expectations are disturbed that we commence to wonder and to investigate. The character of a sound informs us of its probable source and time of origin. We know what to expect when we start out to identify that source and we have but to trace it back to find it.

The recording medium introduces a new element—time. The reproduced sound may no longer be traced directly to its source in point of time. Any period may elapse between the original inception and the final reproduction. The identification of source becomes a voluntary effort, and a multitude of questions arises to perplex us in the technique of the reproduction system. The motion picture craftsman desires to create in his product an illusion which plays upon the imagination of his audience to make them forget these artificial factors. By the many artifices at his command he may often transport them from their own

sphere to the entirely new surroundings which he provides for them at the screen.

1. THE RECORDING SYSTEM A ROBOT

The ear is but a part of the sense of hearing. It is the means of transmitting outside sound phenomena to the brain where the final impression is made. The brain selects that which is wanted and discards the rest so that, within reasonable limits, we hear only that which we wish to hear. This faculty of selection and concentration is very valuable to us in the atmosphere of noise which is always around us. It makes conversation possible in the midst of crowd noise at a football game. Again, in quiet surroundings, we can select from among several persons all talking together, the voice of the one we wish to hear. Physically the presence of our two ears—the binaural sense of hearing—is an invaluable aid to such concentration. Sounds reach the two ears with minute differences of time and phase and loudness which give us a means of identifying the location of the source. Perspective is established and the world of sound becomes three dimensional.

A microphone and recording system have no such discretion. They are robots which pick up everything within their range and record it to the best of their ability. They possess no inherent means of selecting only that portion which is wanted. Emphasis is not possible to them without human guidance. Moreover, the recording system is, up to the present time at least, a monaural or single-eared arrangement. Binaural reproduction is practicable experimentally and should be an advantage when perfected for general use. In any case the direction of the robot, the provision of a brain for the microphone, devolves upon the sound man. He must control its location, directional qualities and response so that the audience will hear only that which is intended for it. Reduction of undesired extraneous noise to a minimum, exaggerated perspective, control of loudness and character of sound are in his hands to provide an illusion of pseudo binaural hearing and realism on the screen.

2. NATURALNESS OF ILLUSION

Sound must naturally just fit the picture on the screen and it must seem to come from the apparent source in the picture. The philosophy of the sound illusion is complex, but fundamentally it is largely a matter of perspective and loudness. Perspective is built into the record with theatre reproduction conditions in mind. Loudness, however, is susceptible to adjustment in both the record and theatre and is a study in itself.

An audience clearly demands that more be heard from a motion picture than from the usual stage presentation, apparently recog-

nizing the voice limitations of the actors on a stage and being willing to accept the loss of certain parts of the show which cannot be heard. The motion picture audience is more difficult to please as it demands complete understanding of every word. Inasmuch as the timing of audience reactions must be predetermined and built into the picture, this demand is often hard to meet. Fortunately the correct loudness or sound volume to fit the picture is invariably greater than that which would be heard from a person at the screen, as the screen figure is usually larger. In passing, it is also of importance to note that all people do not hear equally well, which means a compromise adjustment to fit the majority.

A person raises his voice to speak more loudly. The loudness change is created by a physical change in the voice characteristics. The pitch increases, the power content becomes located higher in the frequency range and the bass elements are reduced. Somewhat the same conditions hold for any noise or for music, with the reservation that the pitch change may be either less or not present. With a reproducing system. however, loudness may be artificially adjusted over a virtually unlimited range. In such an adjustment, the entire frequency range is raised or lowered an equal amount. If the volume is improper and does not correspond to the original, the ear hears an unnatural sound from the distorted relation between the various elements. Too-loud reproductions sound over-bassed, while thinness and lack of bass result from too-low volume.

The recorder attempts to have his actors speak at a voice level appropriate to the scene and surroundings. To make this voice seem natural at the screen after its loudness is increased to fit the picture, he then adjusts the gain-frequency response of his system—equalization—to compensate for the expected difference in loudness. If the actor fails to speak in the proper tone of voice, further equalization is required to make the final reproduction sound right. If, after these corrections, the theatre plays the sound at improper volume level or loudness, the whole balance is further upset in spite of the fact that the record was properly made.

Loudness is greatly affected by reverberation. The ear hears both the sound coming directly from a source and a series of reflections in various relations of phase and attenuation. Recording technique must not only recognize reverberation but must employ it as one of the factors to enhance the illusion. One effect of monaural listening is an apparent increase of reverberation over the binaural condition. Hence a general reduction of the normal reverberation is necessary in the record. A departure from this rule may often be desirable for certain forms of

musical recordings. In every case the original reverberation on the set must be controlled and used with reference to the additional reverberation which will be introduced by the theatre acoustic condition.

The nature of the overall gain-frequency characteristic of a complete system has always been the subject of much discussion. On first thought it would appear that distortionless reproduction would be best obtained by a system having every element capable of uniform response at all frequencies. This would be perfectly true were it not for the necessity for the adjustment just mentioned, to compensate for louder reproduced volume. There is fairly common agreement upon the nature of this adjustment, which is in general an attenuation at the low frequencies. In addition the agreement is becoming more general on the total range of frequencies which should be accommodated. About 50 to 7,500 cycles is more or less accepted as the useful range, with a definite feeling that even the upper end of this range may at times cost more in apparatus refinements and difficulties than its reproduction may be worth.

3. RADIO VS. PICTURES

The sound technique for motion pictures is quite different from that of radio. Radio music and speech require nothing of the eye. The listener may create his own imaginary picture of the setting, or more often probably gives it little or no thought. Moreover, he adjusts the volume to his immediate pleasure with no reference to the probable original volume or to any illusion of the source having been in the room. In the radio studio no limitations exist to control the microphone placement—there are no lights or camera lines to be avoided. A psychological factor is that radio is more often used as a general background with a lowered concentration demand upon the listener. The motion picture illusion, involving both sight and sound, must be more perfect and its requirements are much more complex. Aside from the need for perfect synchronism between picture and sound, the similarity of apparatus and transmission methods is recognized, but the philosophy of their use in motion picture work is different.

4. AN ENTERTAINMENT FACTORY

The motion picture industry is from every point of view one of the most peculiar in the world. Regardless of artistic desires which are its foundation, the commercial end is of manifest importance. Occasionally an artistic effort is made with no thought of pecuniary gain but it is usually the case that a profit must be made. This requirement is sometimes difficult to meet in terms of the purely artistic. The business is one of increasing demands upon the technical staff to maintain the perfection of past performances and to continually be developing

and improving the technical processes to give greater freedom to the creative genius of the producer and his staff.

Entertainment value is a measure of the success of the production. Entertainment will, however, be found to be a very complex and clever combination of all of the many elements available to the producer. The motion picture studio may very aptly be termed an entertainment factory. In many respects it is organized on the same principles as those followed in any other type of manufacturing plant. The raw material happens to be stories and situations, actors, music and effects. The embellishment and processing which this raw material gets is the technical treatment at the hands of the art director, cameraman, sound man, musician, editor and many others, guided and moulded by the hand of the director who is a master craftsman and designer with full knowledge of the capabilities and limitations of his tools.

An important corollary is that all of the various people working with the director must not only be thoroughly familiar with his desires and aims but must also have a well rounded experience and knowledge of the work of the other departments. The set designer cannot disregard acoustics even though he be tempted to use a particular form of construction for some other reason. The camera and sound people have the constant problem of the relation between camera angles and microphone location, affecting both perspective and interference between the camera lines, lighting and microphone. The musician designs his music and lyrics for the scene. Scenes are made and remade for perfection and scores of thousands of feet of film are shot. Finally the editor draws all of this material together into final finished story form and must depend upon the work of the others who have made the film to furnish him with material which has the proper mechanical and dramatic elements to combine properly into a dramatic entity of suitable length to be released.

Probably the simplest thing which the sound man does is to record sound. His familiarity with all of the capabilities and limitations of his sound system as such is the familiarity of a workman with his tools. His real creative work involves complete understanding of the overall showmanship problem and of the interlocking requirements of the several crafts. It happens that his own technical language is that of decibels, gammas and equalizers which are entire strangers to the rest of the motion picture world. It also happens that the sound man's training has usually been such that he is able to assimilate the basic knowledge of the other fields and can meet them on their own ground. The reverse is often not true. It is unfortunate that the sound man may have the reputation of being "high brow." Often he appears to ob-

struct the way of progress because he alone recognizes some of the limitations beyond which, at the moment, it is not practicable to try to go. It should be recognized that this same sound man is well able to plan the details of a scene and to produce the desired overall effect while still keeping within the limitations imposed by physical laws which no one has yet found a way to circumvent. This is not a plea for the sound man but merely an observation of experience, an observation which, when followed, usually has a beneficial result upon the completed productions.

5. A BRAIN FOR THE MICROPHONE

The sound man is the brain of the microphone. He guides the technical system and its action to produce the illusion which is desired on the screen. Without him many undesired sounds which actually occur on the set but which have no bearing upon the production might easily be recorded into the finished product. With him the desired selection and emphasis are obtained. Adjustments to compensate for the monaural listening ability of the microphone are secured by his apparatus control so that he records only that which is wanted, discarding all undesirable material. His technical knowledge of the mechanics of acoustics, equalizers, filters and other mechanisms peculiar to sound work is brought to bear. It is his job to handle and combine a series of objective factors to produce a subjective result which will entertain.

In most instances the sound man is a whole department. It is the practice, particularly in the major studios, for the control of the sound policy to be maintained within certain limits. General type of quality and characteristics, specification of acoustic and other working conditions on the set, laboratory control, maintenance of a complete electromechanical recording system—these and a multitude of other factors are coordinated and controlled by a sound department as such. The individual recordist may then go out on the firing line—the set—with a system at his command which is carefully maintained and adjusted for him and he is free to work out and to coordinate the many details which arise there. Far from stifling the individual and his efforts, this mode of operation gives him more freedom because he is relieved of the care of many technical matters which can be designed and maintained for him.

There is another member of the sound fraternity who is not usually included in a discussion of recording and reproducing technique but who plays a major part in the success or failure of the studio's sound efforts. This is the theatre projectionist, or it may be the management. Whatever the studio sound may be when reproduced under carefully controlled studio conditions, the final reproduction is in the theatre

and under its control. It is gratifying to know the degree of favorable results and intelligent handling in this essential part of the field even though much remains to be done in modernizing equipment in many theatres. This is probably the most important technical problem confronting the sound end of motion pictures at the present time. Fortunately it is being approached in a manner which indicates that improvement may be expected.

6. THE TECHNICAL SYSTEM

The basic elements of a sound recording and reproduction system are perhaps fairly well understood by even the layman. However, a brief review of the fundamental processes may not be amiss, to provide the reader with some form of outline into which he may fit the more detailed discussions as they appear.

Sound recording and reproduction involve a series of transformations of energy from one form to another. The listener should hear by means of the recording the same sound he would have heard had he been present when the original was created. If a simple reproduction system is interposed between source and listener, the latter hears the sound through the system simultaneously with its creation. The further interposition of a recording mechanism places the sound in permanent form on a recording medium so that it may be reproduced at any future time. Any such system can do no more than faithfully reproduce an original. Actually all systems have certain inherent limitations and their faithfulness is measured by the degree of refinement attained in their design and use.

Sound emanates from a source in the form of pressure variations in the air. The amplitude, frequency and phase relations of these variations are determined by the character of the sound. Their effect upon the ear produces the sensation of the particular sound. The recording and reproduction system should faithfully maintain the original pressure variation relations or the listener will not hear a counterpart of the original.

The elementary form of the sound wave is the familiar sine wave. In passing, two of its important features should be noted: First, that sound is made up of highly complex and constantly shifting combinations of frequency, relative amplitude and phase relations; second, that any wave form or group of wave forms can be shown to be the result of a combination of a number of simple sine waves of various frequencies and other determining characteristics. Not only is this true of sound but it also applies to all of the electrical, mechanical and other systems which appear in the recording process. It should be further noted in this connection that a relatively sharp distinction exists between tran-

sient and steady state conditions. These impose entirely different demands upon the technical system, even though they may both be reduced to a sine wave series.

The diaphragm of a microphone placed in the field of a sound wave will vibrate in sympathy with it, resulting in mechanical energy in the form of the diaphragm movement. Microphones of any type depend upon the equivalent of such diaphragm movement to change the resistance of a carbon button, to vary the capacity of a condenser, to move a coil or ribbon within a magnetic field or to modify the pressure upon a piezo-electric crystal. Any type of microphone thus effects a change in energy from the mechanical form of diaphragm movement to an electrical form as the output of the instrument.

In electrical form the energy variations are in such condition that any desirable amount of amplification may be provided by means of the conventional vacuum tube amplifier. Moreover, it becomes practicable in this stage to make any changes or corrections which may be desirable in relative frequency response or in other characteristics of the signal. Too much cannot be said of the importance of the amplifier and vacuum tube to sound work. As in radio and other allied fields, the vacuum tube amplifier makes the whole process practicable.

With the electrical signal amplified to a sufficient degree, the next steps are further transformations into mechanical, and finally chemical form, using light in the transition, through the medium of the modulating device which regulates the signal that goes on the film. The modulator device is in general an electrically operated mechanical shutter of which there are many types. It controls the light from a constant light source in such a manner that the amount of light falling upon a continuously moving recording film varies from instant to instant in proportion to the signal given to the modulator. The film thus receives a varying exposure which is in practically strict proportion to the original sound wave. In the ladder-like striations which appear on a variable density film record, the degree of density or film transmission change is a measure of loudness, while the length or duration of the striation is a function of frequency. With the film moving at a velocity of 90 feet per minute—18 inches per second—each cycle of a 1,000-cycle wave on the film is eighteen one-thousandths of an inch long, whereas a 6,000-cycle wave obviously is but one-sixth of this length. The variable area striation expresses relative loudness in terms of varying width on the sound track—again a variation of transmission. The striation duration or length must, of course, be the same as for variable density. Suitable chemical processing of this negative film record, that is, development and printing in much the same manner as for any

photograph, produces a positive film record capable of reproducing the signal source. This record may be reproduced at any time, and, combined on a single film with the picture to which it applies, may be used for theatre projection.

7. REPRODUCTION

Reproduction is a further series of energy transformations. The first step is optical and electrical. A fixed light source is focused to a sharp line on the film record while the latter is moving in front of a photo-electric cell. The cell thus sees varying amounts of light through the moving film record and has the important function of translating them into corresponding electrical variations. Once more the signal energy is in electrical form which can again be amplified to the desired degree and led into specialized types of the conventional loud-speaker.

The loud-speaker has an electrically operated diaphragm, coupled to the air through some form of horn or projector. The diaphragm is actuated by the amplified electrical energy from the photo-electric cell, and in vibrating mechanically produces in the air a series of pressure variations. If the process has been carefully carried out we are back where we started. These pressure variations will be accurate counterparts of those which emanated from the original source and the listener will hear sound which is a virtual copy of the original and which can be played at any required or desirable volume.

To recapitulate, the several steps in the process and the energy transformations or translations are as follows:

Recording	1.	Source	Acoustic
	2.	Air Medium	Air Pressure
	3.	Microphone Diaphragm	Mechanical
	4.	Microphone Output	Electrical
	5.	Amplification	
	6.	Modulator	Mechanical
	7.	Optical System	Light
	8.	Film—Negative Record	Photochemical
Reproduction	9.	Film—Positive Record	
	10.	Optical System	Light
	11.	Photo-electric Cell	Electrical
	12.	Amplification	
	13.	Loud-speaker Diaphragm	Mechanical
	14.	Loud-speaker Output	Air Pressure
	15.	Ear	Acoustic

The sound man is concerned with each of these translations, their inter-relations and the technical limitations and capabilities of each.

Acoustics, frequency response, harmonic production, phase shift, resonance, power capacity, acoustic efficiency, film characteristics, optics—all of these and many more play their parts in the process and must be controlled. Electrical circuits, amplifiers, photo-electric cells, filters, equalizers, attenuation and transmission elements must continually be combined and employed most effectively for the particular end in view. The sound man must be intimately familiar with the operating principles and available types of microphones, amplifiers, modulating devices, photographic processes, loud-speakers, and the mechanical film recorders and reproducers. His success depends upon his ability to properly handle these various tools and to direct their uses in the best way to secure the result which is desired for the final product. It is the hope that the later chapters of this volume may be of benefit in securing a better understanding of these tools, and of the recording and reproduction problem as a whole and in detail.

Chapter II

THE NATURE OF SOUND

By L. E. CLARK

1. INTRODUCTION

In sound motion pictures, dialogue, sound effects and musical sequences must be picked up and recorded in some permanent manner so that later they may be reproduced in the theatre. Since the recording equipment must be acted upon by the sound, its operation depends upon the physical nature of sound, and upon how the sound behaves between the time it leaves its source and the time it arrives at the receiver. Likewise, the reproducing equipment must be the new source of sound, and what the auditor hears depends upon this physical nature of sound.

Therefore, it is proposed to discuss and describe some of the fundamental characteristics of sound so that application of these properties can be made in later chapters. To be able to do this we must understand how sound behaves; that is, how it is transmitted from place to place, how it is reflected, absorbed, refracted, and diffracted, and what effect it has upon the ear. All these things must be known so that they can be used in delivering to the auditor sound of the same quality that he would enjoy were he present at the original performance.

2. ORIGIN OF SOUND

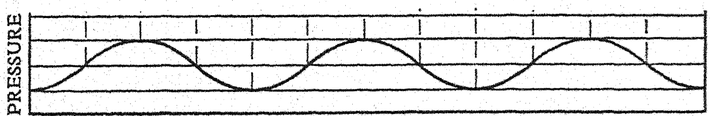
In order to describe these fundamental characteristics of sound it is first necessary to define sound itself. A very good, easily remembered definition is that *sound is a compressional wave motion in an elastic medium.* The American Standards Association defines sound as *"an alternation in pressure, particle displacement, or particle velocity, propagated in an elastic material or the superposition of such propagated alternations. It is also the sensation produced through the ear by these alternations."*

Sound is generated by a vibrating body. If a bell or a tuning fork is struck, it is set into vibration; these vibrations are transferred to the surrounding medium (air in the usual case), which in turn transmits them to the receiver (ear drum, microphone, etc.). Sound

may also be generated by vibrating air columns, as in organ pipes or the human voice, and by vibrating strings. In the case of stringed instruments, the strings themselves transfer very little vibration to the air, but instead vibrate a sounding board or the body of the instrument which transfers much more vibration to the air. In order to transfer enough vibration to the air for the sound to be audible, it is necessary that a considerable area of the vibrating body be in contact with the air. In general, any vibrating body, solid or fluid, will generate sound, but some medium is necessary to transmit the sound to the listener. A bell, ringing in an evacuated bell jar, cannot be heard, although the clapper can be seen to be vibrating. As soon as air is allowed to enter the jar the bell can be heard.

3. SOUND WAVES

As the vibrating body moves forward, the air immediately in front is compressed and the pressure thus built up is relieved by the wave of compression moving outward. As the vibrating body moves back, the air is rarefied, and this wave of rarefaction moves outward behind the wave of compression. Since the vibrating body executes the motion periodically, there will be a train of alternate compressions and rarefactions travelling rapidly away from the body. If the air be examined at any given instant of time there will be found, assuming



SIMPLE PRESSURE WAVE–SINE WAVE

Figure 1.

a simple sine wave of vibration, a series of compressions and rarefactions equally spaced along the direction of travel. The distance between compressions, or rarefactions, or the corresponding points of any two successive waves is known as the wave length, which is designated by "λ." The number of complete vibrations that the body makes in one second is known as the frequency of vibration. Since the body is vibrating with a frequency of "f" cycles per second, it will send out f waves during a second, and as each wave has a length of λ, the sound travels over a distance of $f\lambda$ in one second. The distance traveled in one second is the velocity. Therefore, the relation between velocity, frequency and wave length is,

$$C = f\lambda \tag{1}$$

where C = velocity of sound.

This space picture of the sound wave may be expressed by the equation:

$$p = P_M \cos(-kX) = P_M \cos\left(-2\pi \frac{X}{\lambda}\right) \tag{2}$$

where $p =$ instantaneous pressure above static atmospheric pressure

$P_M =$ maximum instantaneous pressure, or the pressure amplitude.

$X =$ the distance from a reference point

$k = \dfrac{2\pi}{\lambda}$

If at any one point the air be examined over a period of time, it will be found to undergo periodic compressions and rarefactions as the sound passes by. The time between successive compressions is called the periods "T," and is the reciprocal of the frequency, that is,

$$T = \frac{1}{f} \tag{3}$$

The time picture of the second wave is given by:

$$p = P_M \cos \omega t = P_M \cos 2\pi ft = P_M \cos 2\pi \frac{t}{T} \tag{4}$$

where $t =$ time measured from a reference point

$\omega = 2\pi f$

If these two pictures are now combined, the complete space-time picture of the wave can be given by:

$$p = P_M \cos 2\pi \left(\frac{t}{T} - \frac{X}{\lambda}\right) = P_M \cos k\ (Ct - X) \tag{5}$$

This is strictly true only for plane waves, some modification being necessary for spherical and other types of waves.

The sound pressure, "p," is usually measured in dynes per square centimeter. An idea of the magnitude of a dyne can be had from the fact that a mass of one gram weighs 981 dynes. Normal static atmospheric pressure (14.7 lbs./sq. in.) is of the order of 1,013,000 dynes per square centimeter. Ordinary sound pressures range from a fraction of one dyne to several dynes per square centimeter, the threshold of a normal ear being taken as 0.000204 dynes per square centimeter at 1,000 c.p.s. Very intense sounds will have pressures of 100 or 200 dynes per square centimeter. It is seen, then, that the range of pressures in ordinary sounds is very great, of the order of 10^6 to 1, or 120 db, and yet this variation in pressure is very small compared to ordinary atmospheric pressure. The fact that these low level sounds can be heard is due to the remarkably high sensitivity of the ear.

4. VELOCITY OF SOUND

Sound travels away from its source with a velocity that depends upon the medium which is carrying the sound wave. The heavier the particles of the medium or the denser the medium, the more slowly will the wave travel, and the less compliant or more elastic the medium, the faster will the wave travel. The velocity is related to these characteristics of the medium by the equation,

$$C = \sqrt{\frac{E}{\rho}} \tag{6}$$

where $E =$ elasticity

$\rho =$ density

For air at 0° C., $E = 1{,}400{,}000$ dynes/sq. cm.

$\rho = 0.001293$ grams/c.c.

$C = 33{,}100$ cm./sec. or 1,080 ft./sec.

The velocity of sound in some of the more common materials is given below:

air (15° C.)	1,120 ft./sec.
water (15° C.)	4,700 ft./sec.
steel	16,500 ft./sec.
brick	12,000 ft./sec.

Because the elasticity and density vary with temperature, the velocity will also vary with temperature. The velocity of sound, however, is not dependent upon the frequency over the range of frequencies generally referred to as audible sound. It is not dependent upon the intensity except for very high intensity sounds, such as a thunderclap, in which case there is some increase in velocity.

5. REFLECTION OF SOUND

When sound traveling in one medium encounters a change in that medium or encounters another medium (such as water, colder air, or a solid object), which is large compared to the wave length of the sound, it is partially reflected back into the first medium, and partially transmitted by, and absorbed in, the second medium. The same law of reflection applies as in the case of the specular reflection of light, i.e., *the angle of incidence is equal to the angle of reflection.* In Figure 2, sound traveling in the direction AD, strikes the boundary of the two media, OO′, at an angle of incidence ADC and is partially reflected in the direction DB, at an angle of reflection CDB equal to ADC. The relative amount of energy reflected depends upon the densities of the two media, and upon the velocity of sound in them. The ratio of intensity, or

energy, of the reflected sound to that of the incident sound, is the acoustic reflectivity, "r."

The energy not reflected is transmitted by the second medium or absorbed in it. In the use of acoustical materials for sound absorption, which will be referred to later, we are interested in how much of the sound energy will be removed from the room. Since this represents all

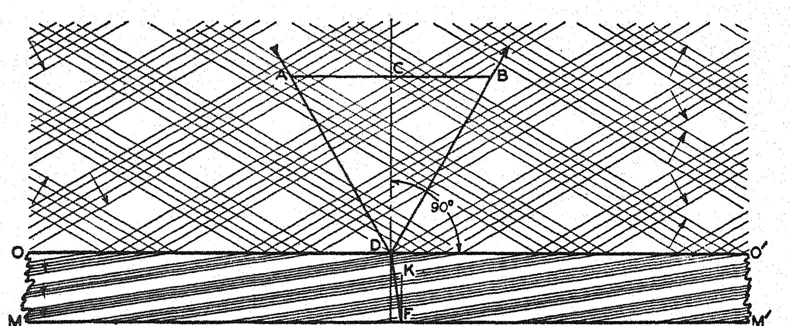


Figure 2.

the energy except that which is reflected, the acoustic absorptivity is defined as one minus the acoustic reflectivity, that is, $a = 1 - r$.

For reflection to take place, however, it is necessary that the reflecting surface be large—at least several wave lengths—in comparison with the wave length of the sound. If the surface is smaller than this, a phenomenon known as diffraction begins to become effective and it is impossible to locate the reflected sound by the laws of reflection.

6. REFRACTION OF SOUND

Referring again to Figure 2, the sound wave in the second medium travels in a direction different from that of the incident sound. This phenomenon, which is called refraction, is due to the difference in the velocity of sound in the two media. As the velocity is less in the second medium, the edge of the wave slows up when it enters this medium. The part of the wave remaining in the first medium is still traveling with its initial velocity and as a consequence there is a shift in the direction of travel from AD to DF. If C is the velocity in the medium and C′ that in the second medium, then

$$\frac{\sin \angle ADC}{\sin \angle KFD} = \frac{C}{C'} \tag{7}$$

The effect of refraction is most often observed when it is due to the existence of layers of air at different temperatures, such as cool air over a warm body of water in the evening.

7. DIFFRACTION

Diffraction is the bending of sound waves around an obstacle. If a window is opened only slightly, the entire room will be filled with any sound present on the outside. However, sunlight coming through the window, instead of spreading throughout the room, will cast a sharp shadow. If the edge of the light shadow is closely examined, diffraction would also be found, but on a much smaller scale than in the case of the sound wave.

This difference in the amount of diffraction present in the two cases is due to the different wave lengths of light and sound; sound waves being several feet in length and light waves being some twenty millionths of an inch in length.

Diffraction effects can be thought of as existing in a narrow region, only a wave length or so in extent, at the edge of the wave. Thus, when a sound wave passes by a large obstacle, such as a building, most of the energy goes straight on and only a small part is diffracted into the shadow caused by the building. However, when sound strikes a window which is open but a few inches (only a fraction of a wave length), the sound spreads out as if coming from a point source. Since most objects around us are only several wave lengths in size, we are accustomed to diffraction effects and usually do not give a second thought to the fact that we often listen to sounds whose source we cannot see.

8. INTERFERENCE

When two waves of the same frequency combine, they give rise to an effect known as interference. At points where the two waves execute their vibrations in-phase, the maximum pressure will be the sum of the pressures in the individual waves, while at points where the vibrations are 180° out-of-phase, the resultant pressure will be the difference in the two pressures. The sound will, therefore, be re-enforced at some points and destroyed at others, giving an irregular distribution of sound energy. Interference can be caused by waves coming from more than one source or by the combination of the direct and reflected sound from a single source. This commonly happens in theatres and studios, and accounts for the dead spots often encountered at certain frequencies. The interference pattern can easily be detected by establishing a pure tone in a room and then listening to the tone at various points in the room.

9. ARCHITECTURAL ACOUSTICS

The acoustics of the recording studio and theatre are of great importance because these rooms constitute two links in the chain from

the original source (through the recording and reproducing equipment), to the auditor, and any distortion originating in these rooms will be passed on to the listener. While modern science and technical advancement have made possible recording and reproducing systems that introduce negligible distortion into the sound, there is still opportunity for distortion to enter between the source and the microphone and between loud-speaker and auditor. A listener in a room seldom hears sounds that have not been changed in character in some manner en route to him. The various frequency components of the sound will be acted upon differently by the same absorbing materials in the room, the low frequencies usually being absorbed less than the high frequencies.

Some surfaces are of such size that they will reflect the high but not the low-frequency sound, the latter being diffracted around the surface. Often, therefore, a room may be very dead at some frequencies and very live at others, while at still other frequencies there may be detrimental echoes. In addition to distortion produced in this manner, the studio or theatre may be resonant at certain frequencies, or the structure may vibrate.

These are examples of distortion known as frequency distortion, in which the different frequency components are not all acted upon in like manner. There is also a type of distortion known as non-linear distortion, which is caused by failure of a system to respond in a proportionate amount for different inputs. This distortion, negligible for all but very intense sounds, is present in the air at all times because air follows the action of a gas under pressure.

As a result of the effects of serious distortion, sound usually becomes unnatural, hollow or unintelligible.

When an observer listens to a source of sound in the open, he hears only the direct sound, and as he moves away from the source the intensity of the sound falls off with the square of the distance. If, however, he listens to the sound in a room, the intensity does not decrease as rapidly as he moves away; in fact, it decreases very irregularly and at times may even increase because of the interference pattern of the enclosure. This occurs because he is receiving not only the direct sound but sound that has been once, or twice, or even many times, reflected from the surfaces of the room. To consider the matter from another point of view, when the sound energy is once introduced into an enclosed space, it remains there until used up or dissipated in some manner. Some energy is used up at each reflection and so the total is constantly decreasing.

The time required for the sound in any enclosure to die away to one-millionth of its initial intensity, or over a 60 db range, is known as the reverberation period of that enclosure, and depends upon the volume of the enclosure and upon the amount of absorbing material within it. The energy is used up by any materials that absorb sound which may be in the enclosure, and in the clothing of persons in the room—very little of it being dissipated by transmission through the air.

The sound reaching an auditor in the room consists of the direct sound, the first few beneficial reflections coming from surfaces near the source, and reverberant sound. If a reflecting surface is large and some distance from the original source, the reflected sound will arrive sufficiently later to be recognized as a distinct repetition. This is called an echo. If the reflected sound arrives 1/20 of a second later than the direct sound, there will be a noticeable echo. The reverberant sound is due to the many repeated reflections from the surfaces of the room, and is very detrimental to the perception, causing a hanging-over and general jumbling of the sound. In the case of speech, one syllable may be still reverberating in the room when the next one is spoken, making the spoken words unintelligible. The reverberant sound must be reduced if the room is to be satisfactory for speech or music, and for this reason, auditoriums, theatres and other similar places must be fitted with sound absorbing materials that reduce the reverberation time to a satisfactory value. The optimum reverberation time depends upon the volume of the room, upon the purpose for which the room is to be used, and upon the frequency of the sound. Figure 94 shows the optimum reverberation time for theatres of different size, and Figure 95 shows how this time should vary with frequency for any one room.

Moving picture auditoriums should be somewhat less reverberant than the average auditorium of the same size because the sound originates from the reproducing system, which has ample power, and it is not necessary to rely upon reverberant sound to build up the loudness. Then, too, some reverberation is recorded and this increases the effective reverberation in the theatre. Theatres should be designed so that there are no reflecting surfaces to cause echoes, and no curved surfaces to cause concentrations of sound or dead spots in the house.

In general, recording studios should be even less reverberant than theatres of the same size. This is necessary because the sound is picked up with a microphone and the effect is that of listening with one ear. This greatly increases the apparent reverberation time and the difficulty of making the direct sound stand out above the reflected sound. The ratio of direct to reverberant sound is of great importance to recording engineers because upon it depends the quality of the recorded sound. If

this ratio is too low, the sound seems reverberant, and if too high, the sound seems lifeless.

One exception to the above statement is in studio sound stages used for music recording. In this case a higher reverberation time—compared to that used for dialogue recording—is desirable to reduce the masking effect of music over dialogue.

In the presentation of a motion picture it is desirable to have the magnitude of the sound comparable to the comparative size of the actors in different scenes. That is, the audience expects to hear dialogue considerably louder when it is associated with a close-up or medium shot than in a long shot, or if the person were actually on the stage. For the same reason, sound effects must be exaggerated in like manner. This increase in loudness with respect to the original intensity is, in many cases, greater than 10 db. Because of the shift in characteristic with loudness (see Figure 4), careful control of all these factors is necessary in order that the reproduced sound seem natural, even though it may be louder than the original.

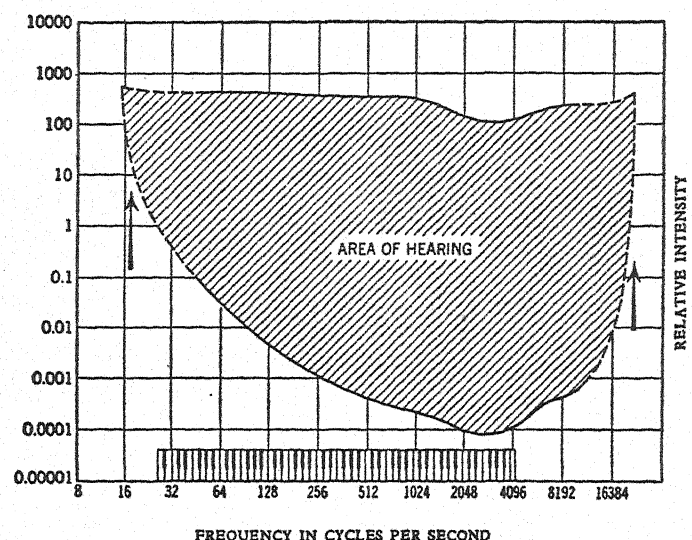
10. PERCEPTION OF SOUND

We are interested in the nature of sound and its effect upon the ear because ultimately sound is to be listened to, and what is perceived by the brain will therefore depend upon the behavior of the ear. Many extensive fundamental investigations have been carried out in this respect by Dr. Harvey Fletcher of the Bell Telephone Laboratories, with the result that our understanding of the process of hearing is very much advanced over that of a few years ago.

The three physical characteristics of sound are: Frequency, intensity and overtone content. Frequency, as previously mentioned, is the number of vibrations made in one second. The intensity is proportional to the amount of energy in the sound wave, and the overtone content is a function of the number, magnitude and phase of frequencies other than the fundamental. Most sounds are not pure tones, but are complex waves consisting of a fundamental tone (establishing the waves on the musical scale), and a number of components of higher frequencies. These latter constitute the overtones. A violin and a cornet, both playing middle C, sound different because the overtones are not the same for each instrument.

The ear does not respond uniformly to all frequencies, but is more sensitive to frequencies of about 1,000 c.p.s. (This is a form of frequency distortion, which is present whether direct or reproduced sound is under consideration.) The ear is able to respond to frequencies ranging from 20 to 20,000 c.p.s. Figure 3 shows the area of hear-

ing of an average ear. The lower line is known as the threshold of audibility and shows the minimum pressure at each frequency that



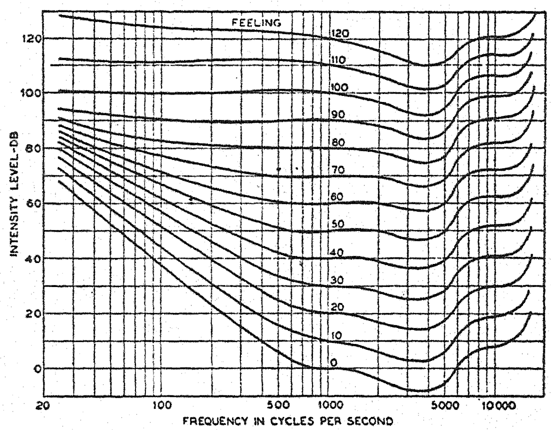
—*From "Sound," by Lemon & Schlesenger (The University of Chicago Press).*

Figure 3 — Recent data on the limits of audibility.

will cause a sensation of sound. The upper line is known as the threshold of feeling and shows the pressure, at each frequency, that will give rise to a sensation of pain. Any sound whose pressure and frequency fall within the area bounded by the lines will cause the sensation of sound.

There are three physiological characteristics of sound which approximately correspond to the three physical characteristics named above; namely, pitch, loudness, and quality. Pitch has been used interchangeably with frequency, but recent measurements indicate that the sensation of pitch also depends to a slight degree on the intensity and overtone structure. The loudness of a sound is approximately equal to the logarithm of the intensity, and is so defined for 1,000 c.p.s. Departures at other frequencies, especially the lows, are considerable and the definition should not be taken too literally for frequencies other than 1,000 c.p.s. Figure 4 shows contours of equal loudness obtained by plotting, on the hearing area of Figure 3, the locus of all points which a large number of observers indicated were of equal loudness. It will be noticed that these contours are not equally spaced but generally converge toward the low-frequency end. This is the cause of the change in quality of reproduced sound when the volume level is changed. For example: Consider a system carrying two pure tones of equal loudness, say 1,000 c.p.s. and 100 c.p.s. and let the 1,000 cycle

tone have an intensity level of + 80 db. Then, reference to Figure 4 shows that the equally loud 100 cycle tone has an intensity level



—From a technical paper in Journal Acoustical Society of America, Volume V, No. 2, "Loudness, Its Definition, Measurement and Calculation," by Fletcher and Munson.

Figure 4 — Loudness level contours.

of + 83 db. If the system gain is now lowered 40 db, the intensity of the 1,000 cycle tone is + 40 and that of the 100 cycle tone is + 43. Again referring to Figure 4, it is seen that the two tones are no longer of equal loudness, there being an apparent 20 db difference between the two. The balance is therefore completely destroyed, as the sensation received by the brain is entirely deficient in its low-frequency tone.

Investigations have shown that high frequencies, that is, frequencies above 1,000 c.p.s., are necessary for the intelligibility of speech. They also give brilliance to the sound. The low and bass frequencies are necessary to retain a material quality to voice and music, and in addition give "presence" to the scene on the screen.

The ear is also subject to non-linear distortion at high-intensity levels, its mechanism being unable to respond faithfully to very intense sounds. It is never possible to listen to undistorted sound as we cannot dispense with the ear between the sound and the brain. In view of the fact that the ear will respond to pressure changes of over a millionfold, and to frequencies over a range of 10 octaves, it must be considered to be a well designed mechanism.

The effect of age upon the hearing ability of the ear is shown in the table below, which indicates that, with age, the ear loses its sensi-

tivity to the high frequencies—the effect lessening as the frequency decreases.

DB Loss in Hearing with Age

Frequency	*60 to 1024 Cycles*	*2048 Cycles*	*4096 Cycles*	*8192 Cycles*
Ages 20-29 (96 ears) - - - -	0	0	6	6
Ages 30-39 (162 ears) - - -	0	0	16	11
Ages 40-49 (84 ears) - - - -	0	2	18	16
Ages 50-59 (28 ears) - - - -	0	5	30	32

It is not generally recognized that natural sound which we hear always emanates from its proper source, whereas in artificial reproduction we usually hear sounds which do not emanate from their original source. That is, in a motion picture reproduction, all sound originates from a fixed source—the speaker system. This sound includes both the sound which seems to originate from all portions of the screen, as well as sound that has its apparent source off the screen. As a consequence of this unnatural source of sound, the many factors affecting the relative loudness of the various reproduced sounds must be carefully controlled.

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Chapter III

TYPES OF FILM RECORDING

By L. E. CLARK *and* JOHN K. HILLIARD

1. RECORDING MEDIA

Sound recorded for motion pictures is of two general types, (1) sound-on-disc, and (2) sound-on-film. At the present time practically all commercial releases (in the United States) are on film, discs being used only within studios for sound playbacks, cueing, etc.

Because of the very limited use of disc recording at the present time, only the recording of sound on film will be discussed in this book.

2. TYPES OF FILM RECORDS

There are today two types of sound-on-film recording in general use: (1) "Variable Area," and (2) "Variable Density."

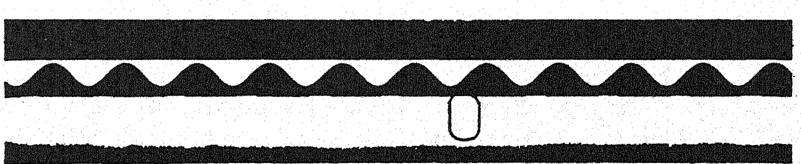
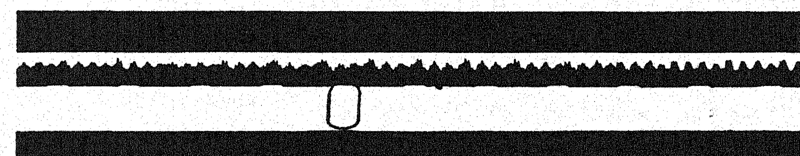
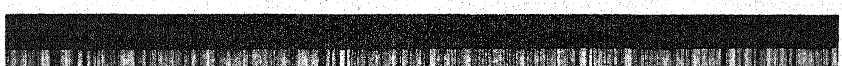


Figure 5-A — Example of a single-frequency variable area recording.



—*Courtesy RCA Manufacturing Co.*

Figure 5-B — Example of a typical variable area recording.



—*Courtesy Electrical Research Products, Inc.*

Figure 6 — Example of typical variable density recording.

In the former type, the record consists of a clear or transparent area adjoining an area of uniform density, the dividing line between the

two areas outlining the wave form of the recorded signal. The variable density record consists of transverse parallel striations of varying density—varying according to the frequency of the recorded signal.

These two types of records may be secured in a variety of ways, but the main distinction between the two is that in one type the area varies while in the other the density varies.

Figure 5-A shows an example of a single envelope variable area recording of a simple tone, while Figure 5-B is a variable area recording of a typical complicated signal. Figure 6 is an example of a variable density recording.

3. METHODS OF RECORDING

In the variable area type of recording only one method is employed for varying the exposure of the film, that is, a beam of light of constant intensity is caused to vary in its length.

In the variable density method, the slit through which the light strikes the film may be held constant and the intensity of the light source varied, or the intensity of the light may be held constant and the slit width varied.

(a) The Aeolight System

This was the first commercial sound-on-film system in use in the studios and gives a variable density record by varying the intensity of a recording light according to the signal, the light striking the film through a slit of constant width (0.001"), positioned between the lamp and the film, which passes the slit at a constant speed.

The aeo-lamp is a gaseous discharge tube, consisting of a nickel anode and a looped filament, designed to give a fairly intense beam of light at the "no-signal" level, accomplished by the use of a direct-current bias as illustrated in Figure 7. The value of the bias varies from 250 to 400 volts.

The lamp circuit shown is connected to the output of the bridging amplifier, the alternating signal component causing a variation in the bias voltage across the lamp and a consequent variation in its intensity.

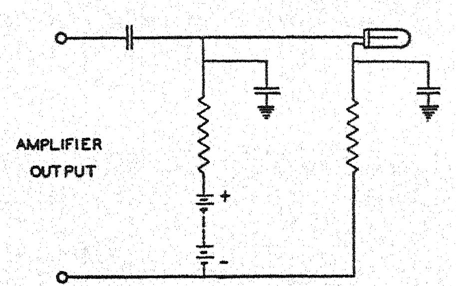


Figure 7 — Schematic circuit of a variable density recording modulator where the lamp intensity is variable.

Figure 8 shows the relation between the light intensity and the voltage. The bias value should be such that the no-signal intensity falls in the center of the straight line portion of the curve, while the maximum alternating-current variation, either plus or minus, must also be within the straight line portion, or distortion will result.

This method of producing variable density records has been superseded in the studios by the variable slit-width method.

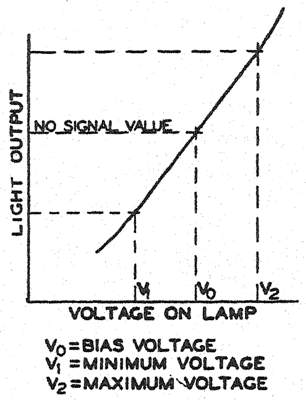


Figure 8 — Curve showing the relation between light intensity and impressed voltage of an Aeo-lamp.

(b) Variable Density Recording

Recording by the variable density method is also accomplished by means of a lamp of constant intensity, the light from which is focused by means of a lens on a slit. This slit varies in width according to the electrical intensity of the signal to be recorded, and so varies the amount of transmitted light in proportion to the electrical impulses. Such a light modulator is called a "Light Valve." The transmitted light is in turn focused upon the film by means of a second lens system.

A current carrying conductor, when placed in a magnetic field, is subjected to a force proportional to changes in the current passing through the conductor. If free to move, the conductor will move a distance proportional to the change in, and the direction governed by, the flow of current. Light valves operate on this principle.

A conductor, placed in a magnetic field and carrying a current, would tend to bow if the ends of this conductor were fixed. The center section of this conductor, if the conductor's length is great compared to its cross-section, would for all practical purposes, move parallel to itself as shown in Figure 9. In commercial application of this principle to the recording light valve, two ribbons are used, as shown in Figure 10.

The light valve ribbon is formed of two strips of duraluminum, 0.0005″ by 0.006″, spaced 0.001″ apart. The ribbon has a natural frequency of its own, depending upon its tension and length, and is consequently tuned for a frequency above the highest frequency to be recorded, which in commercial practice is always above 8,500 cycles. Tuning is accomplished in exactly the same way a violin string is tuned, by changing the tension on the ribbon. The current flows in opposite directions in each of the ribbons of the valve, resulting in an opening

and closing of the aperture between the ribbons when subjected to a signal current.

Originally, the two ribbons were mounted in the same horizontal plane. As previously mentioned, excessive modulation causes the ribbons to "clash" or strike each other. A valve so mounted is called a mono-planar valve.

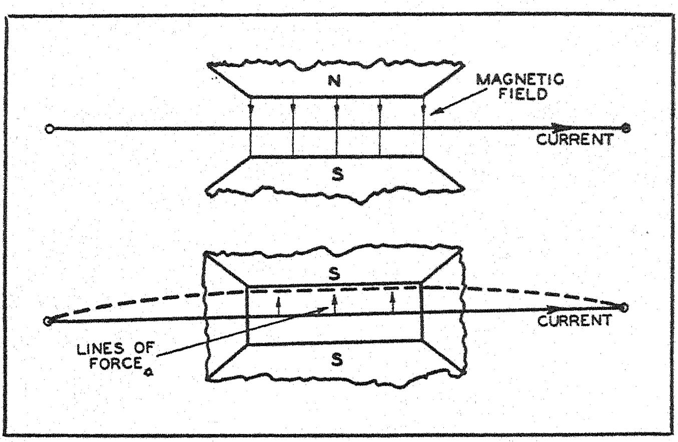


Figure 9 — Diagram showing the light valve operating principle.

In a more recently designed valve, called the bi-planar valve, the two ribbons are mounted, spaced one-half to two mils, one above the other, and vibrate in two parallel planes. This eliminates the possibility of the two ribbons striking together at periods of excessive modulation, resulting in the valve remaining in better mechanical condition, throughout its life.

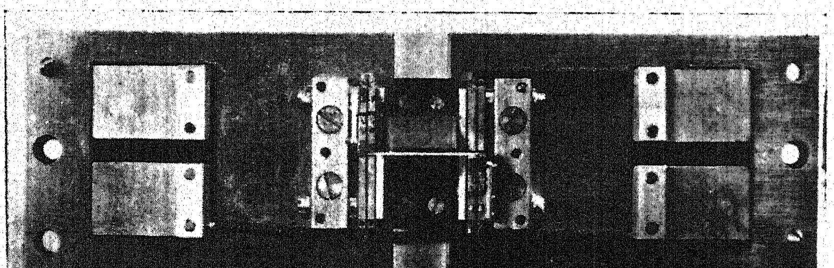


Figure 10 — Two-ribbon recording light valve.

Excellent results have been obtained with light valve recording, and the extensive use to which this method has been put in the studios

shows it to be rugged as well as efficient. However, care must be exercised in its use, and routine checks and adjustments made periodically. Azimuth adjustments must be particularly made, i.e., the horizontal plane of the valve must be perpendicular to the direction of film movement. Focal adjustments must be maintained as it is necessary to photograph the ribbon movements as accurately as possible. "Hysteresis," or the failure of the ribbons to return to their original position, may be minimized by increasing the tension, but recently designed valves are practically free of trouble from this source.

Overload and clash of the two ribbons, which result in distortion of the wave form, should be avoided. These effects occur when the ribbons modulate beyond the full or zero modulation position, which produces either a cutting off of the wave top or an actual clash of the ribbons.

Another source of possible trouble is that of "ribbon-velocity." This is due to the fact that one ribbon when opening and the other ribbon when closing, are traveling in the same direction as the film when it is passing the valve. The effective exposure recorded on the film is not only dependent upon the light flux, or intensity, but also upon the time a given point on the film requires to pass through the slit image; that is, the exposure depends upon the product of light intensity and time. At low frequencies this effect is negligible, as the velocity of the ribbon is small compared to that of the film (normally 90′ per minute), but at higher frequencies the two velocities become comparable and loss of volume as well as distortion in wave shape results.

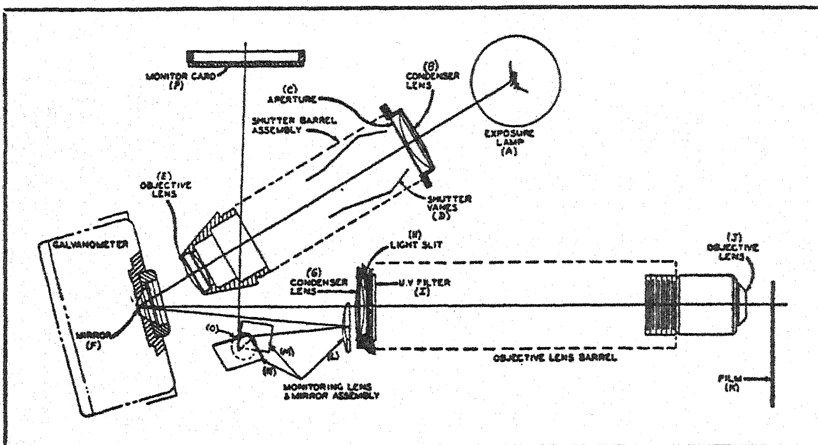
Variable density recording can also be secured by the use of a galvanometer and a penumbra in the optical system shown in Figure 11. This is the optical system normally employed for variable area recording with a galvanometer.

(c) Variable Area Recording

Fundamentally, this system consists of a concentrated lamp, an image of whose filament is focused upon a galvanometer mirror, from which the light is reflected and imaged upon a slit of constant width. The signal currents cause a rotation of the galvanometer mirror allowing either more or less of the slit to be illuminated, and thus is photographed a sound track of variable area, or width, but of constant density. The mean, or "no-signal" position, exposes one-half the sound track variations in the signal current causing a movement of the light beam either way from this mean position.

With this method a more complex optical system is necessary than for the variable density method.

Figure 11 shows such an optical system.



—From the Journal of the S.M.P.E., September, 1937. "The RCA Recording System and Its Adaptation to Various Types of Sound Track," G. L. Dimmick.

Figure 11 — Schematic of a variable area recorder.

The different types of tracks, discussed in the chapter on "Noise Reduction," may be produced by varying the type of aperture used. Single envelope variable area track as shown in Figure 5-B, which was the first used, employs an aperture with a rectangular opening, which at no-signal level is imaged upon the slit as shown in Figure 12 (accomplished by proper bias on the galvanometer). The incoming signal will then vary the position of the mask image upon the slit in proportion to its strength, resulting in an oscillograph record which is opaque on one side and transparent on the other in contrast to the usual oscillogram which shows only as a line on the film record.

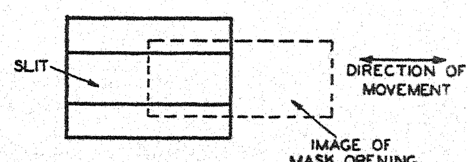
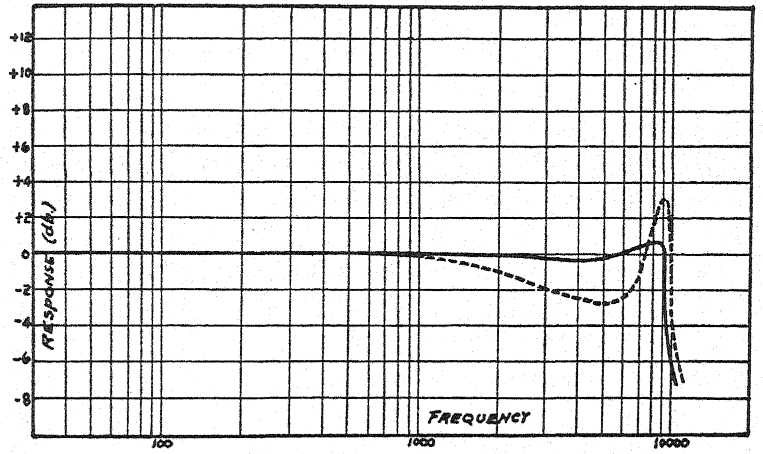


Figure 12 — Type of mask used for standard variable area recording.

The first variable area recording system employed an oscillograph consisting of a pair of fine wires to which the mirror was clamped. These wires were placed between the pole faces of a magnet and operated very similarly to the ribbons of a light valve. The complete assembly was immersed in oil to provide adequate damping. Later galvanometers are

of the moving iron type in which the armature causes the galvanometer mirror to vibrate through a simple mechanical link. These galva-



—*From the Journal of the S.M.P.E., September, 1937. "The RCA Recording System and Its Adaptation to Various Types of Sound Track," G. L. Dimmick.*

Figure 13 — Variable area recording system modulator response with frequency, with galvanometer damped (dotted curve), and with galvanometer damped and with bias condenser (full curve).

nometers are air damped, tuned to approximately 9,500 cycles, and are not critical to temperature changes.

The response with frequency of a modern galvanometer is shown in Figure 13.

The dotted line curve gives the response with no bias condenser in the circuit. The droop that occurs in the high-frequency range is caused by the inductance of the modulation winding. Earlier types of galvanometers obtained their damping by having the unit immersed in oil. Practical difficulties, however, led to the development of a damping assembly made of tungsten-loaded rubber which reduced the resonant peak at 9,000 cycles from about 12 db to about 3 db. Both the droop and peak shown in the dotted curve can be overcome and the frequency characteristic improved by placing a condenser across the bias winding to neutralize the inductive reactance. The effect of the bias capacitor upon the galvanometer response is shown by the full line curve in Figure 13.

With the variable area system certain care is required to obtain optimum results. Obviously, if the galvanometer is over-modulated,

the peaks of the signal will be cut off and distortion will result, while azimuth and focal errors, hysteresis, etc., must be minimized by correct design and adjustment of the galvanometer and lens system.

4. SLIT WIDTH

The physical width of the slit plays an important part in the quality of the resultant recorded sound, especially in the high-frequency band.

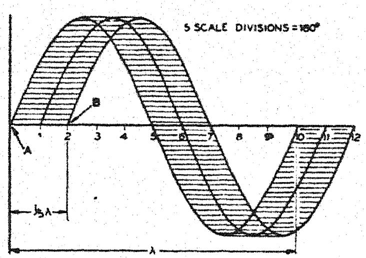


Figure 14-A.

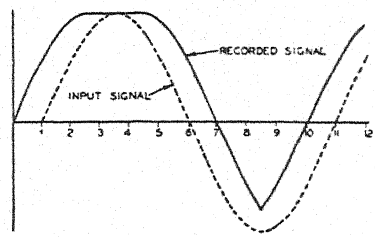


Figure 14-B.

Assume that a wave is to be recorded whose length is five times the slit width. One edge of the slit starts at the point A and the other edge at B as shown in Figure 14-A, with a resulting track as shown in Figure 14-B. It will be noticed that in one case the peak of the wave is lost and in the other accentuated—*but in both cases distorted.*

For this reason, the slit must be narrow in relation to the highest frequency to be recorded.

At 90 feet per minute the film travels 18,000 mils (1 mil = 0.001″) per second, and a 9,000 cycle wave, for instance, would be two mils long. If a slit width of 1 mil is assumed, it can be seen that the

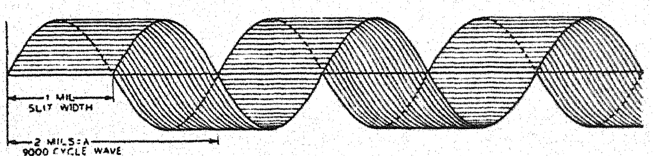


Figure 15-A.

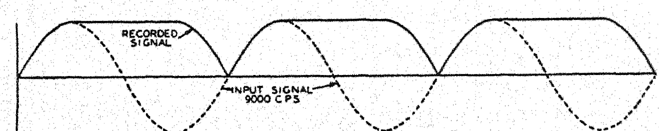


Figure 15-B.

slit width equals one-half the wave length. A study of Figure 15 shows that one-half of the wave is lost, that the other half is distorted, and the intensity changed.

Figure 15-A shows the movement of the top edge of the recording beam in relation to the long dimension of the slit, that is, in this drawing the light beam is shown moving in a vertical direction and the film in a horizontal direction toward the right. The input signal shown by the dotted line curve in Figure 15-B, is a sine wave of 9,000 c.p.s. The resultant recorded signal, shown by the full line curve in Figure 15-B, is the envelope of the area scanned by the recording beam, and shows the effect of slit width on intensity and wave form. With a slit width of 2 mils the response from a 9,000 cycle wave would be zero.

However, the lower limit of slit width is determined by other considerations, principally the difficulty of securing sufficient light to properly expose the film when very narrow slits are employed. The usual practice in the case of variable density is to employ an effective slit of ½ mil and in the case of variable area an effective slit of approximately 2/10 mil. The physical slit itself in both cases is actually considerably wider than these figures indicate, but is reduced by optical means.

5. NEW RECORDING METHODS

The constant quest by the sound departments of all the studios for a practical system of increased noise reduction which does not penalize recording methods or the quality of recorded sound, has led to refinements in the systems discussed above as well as to new recording techniques.

(a) Push-pull Recording

Sound which is to accompany motion pictures must be reproduced with the highest possible accuracy in order that the picture give the greatest possible illusion of reality. In viewing a motion picture, the audience is stimulated to a specific sound expectancy not necessarily present in the radio and phonograph.

In order to approach the ideal illusion of reality it is necessary that the recording and reproducing systems have a frequency response and volume range commensurate with the original. A high degree of linearity is essential. The frequency range should be from fifty to eight thousand cycles and the volume range from fifty to sixty db.

Reproducing systems which will meet these requirements are now available and are being installed in theatres, and the recording practice has been developed to a degree that also fulfills these requirements in the original recording form. However, it has been found necessary to limit the released volume range until such a time as the majority of theatres are equipped to handle the greater volumes.

To date, standard methods of recording on film, both variable density and variable area methods, have given a volume range of approximately forty db, at the expense of considerable distortion. In the variable density system the principal limitation has been the small linear range of density between the toe and shoulder of the characteristic curve for positive stock, plus the distortion added by the noise reduction system because of its relatively slow operating time. Push-pull recording inherently reduces these limitations by the cancellation of the internal distortions.

When sound is recorded on film, either by the variable area or variable density method, certain harmonic and extraneous components appear in the reproduced signal which were not contained in the original signal.

For the variable density recording method these consist mainly of two distortion components; one introduced by the non-linear toe and shoulder portions of the film characteristic, and the second, signal components derived from the noise reduction apparatus. For variable area recording the distortion consists of shutter or bias signals and processing effects.

Push-pull recording, like the push-pull method of operating vacuum tube amplifiers, is a method of reducing certain of these distortion effects in the electrical signal to be reproduced. The similarity between film recording and vacuum tube circuits is quite complete, the one providing a means for cancelling out certain harmonics produced in vacuum tubes because of non-linear tube characteristics, the other accomplishing the same results for film recording in connection with the above mentioned distortion effects. In fact, the possibilities for push-pull recording had their inception in the success of the vacuum tube development.

This type of recording is secured by means of two sound tracks on the film which are reproduced by a double photo-electric cell arrangement, which adds the signals from the two tracks in their proper phase relation. The two half-tracks are recorded out-of-phase on the film and the double photo-electric cell circuit is arranged to have a phase turnover, so that the signal will be reproduced in-phase (see Chapter XXII). This phase turnover is accomplished in the same manner that the outputs of two push-pull vacuum tubes are joined together by a center tapped transformer. Thus, even order harmonic content introduced by the film is cancelled out because of an out-of-phase condition in the combining transformer. As in push-pull vacuum tube circuits, it is not possible to cancel out both even and odd order harmonics.

In push-pull recording, noise reduction signals are recorded on the two half-tracks in-phase in contrast to the out-of-phase "wanted" signal. These former signals will, therefore, be cancelled out in reproduction by the photo-electric cell circuit due to its 180° phase change. Because of this cancellation feature, it is feasible, if desired, to speed up the operation of the noise reduction system (that is, allow the noise reduction signal to contain more frequency components), without damage to the reproduced signal content. This tends to decrease the amount of wave top clipping on steep wave front signals, which is a result of the sluggishness of operation of the noise reduction equipment. (See chapter on Noise Reduction.)

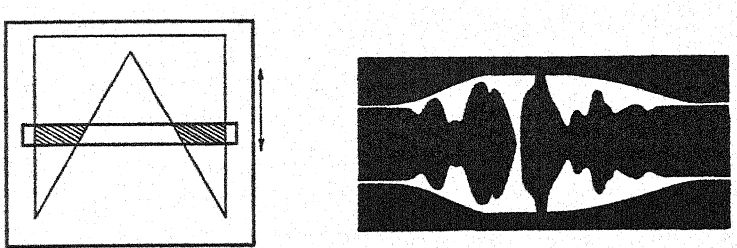


Figure 16-A — Duplex variable area.

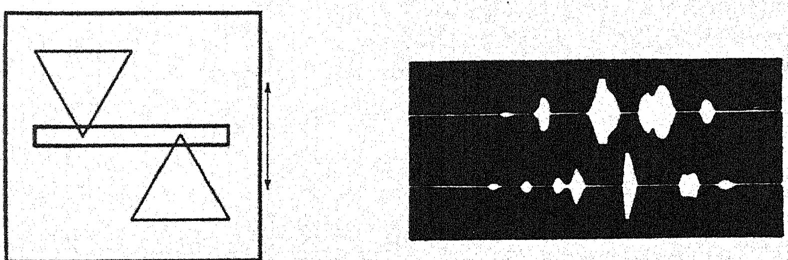


Figure 16-B — Push-pull Class B variable area.

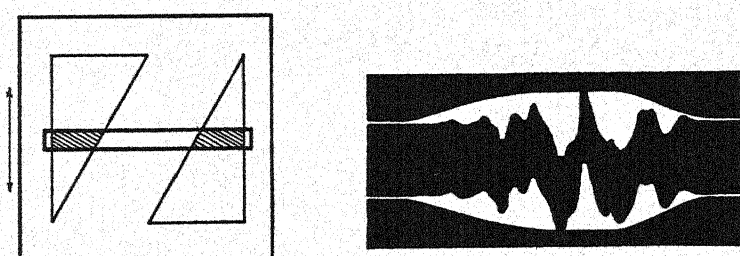


Figure 16-C — Push-pull Class A variable area.

Push-pull recording may be of three types: Class A, Class B, or Class AB—the class notation having the same significance as for amplifier operation.

In Class A recording, half the total signal energy is recorded on each half-track with the result that they are alike but out-of-phase as discussed above. (See Figure 16.)

In Class B recording, one half-track receives the positive signal energy and the other half-track receives the negative signal energy.

The adjustments necessary to produce each of these tracks are accomplished by proper external connections to the light valve in the case of variable density, and by the inclusion of the proper aperture in the variable area optical recording system.

As explained in the chapter on Noise Reduction, Class B track is not suitable, at least at the present time, for general theatre release, because of the necessity of maintaining an accurate sensitivity balance in the push-pull reproducer. Inequalities resulting from the processing of the film also add to the reproduced distortion.

The push-pull, Class A, variable area recording is accomplished by converting the optical system by change of apertures, using the same shutter in both cases. (See Figure 16.)

As the noise reduction signals are cancelled out by the reproducer circuit, increased speed of operation of the shutter, if desired, may be secured. In this case, as in reproduction from a variable density push-pull record, a considerable reduction in even harmonic distortion is obtained and distortions due to audible variations in average print transmission are cancelled out.

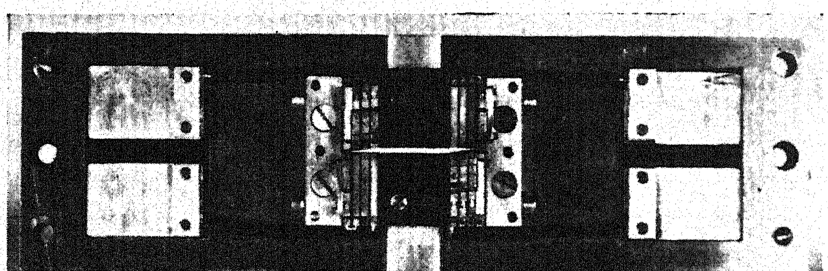


Figure 17 — Four-ribbon, push-pull light valve.

Class AB recording consists in adjusting the recording mechanism to record Class A for low volume signals and Class B whenever the signal volume exceeds a given amount. The point of separation between the Class A recording and Class B recording is determined by the setting

given to the recording mechanism. Class AB recording, up to this time, has been confined mostly to experimental work.

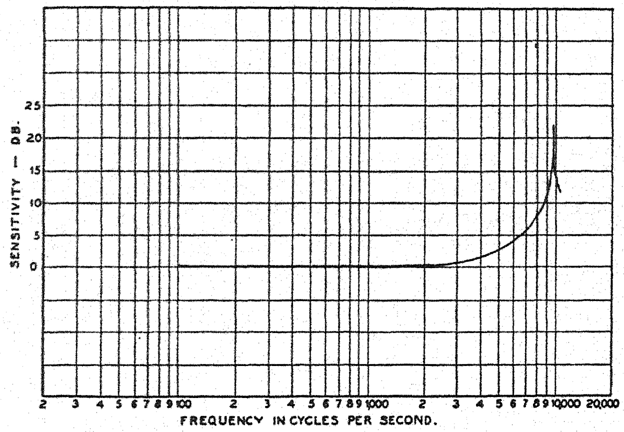


Figure 18 — Sensitivity characteristic for four-ribbon push-pull light valve (ERPI valve-RA-1061A).

For push-pull variable density recording, the light valve has been modified, as shown in Figure 17, to have two pairs of bi-planar ribbons of conventional type, each pair exposing half of the sound track, with the valve so connected that these ribbons act in push-pull. It is mechanically impossible to mount the ribbons in line with each other, so, in order to scan the track with a single slit, the image must be optically relocated on the film. This is accomplished by placing an optical flat in the path of each light beam in the form of a saw buck as shown in Figure 19. This arrangement moves the axis of the path in proportion to the angle at which the optical flat is interposed. As a result, the opposing densities on the two halves of the track are positioned horizontally across the film.

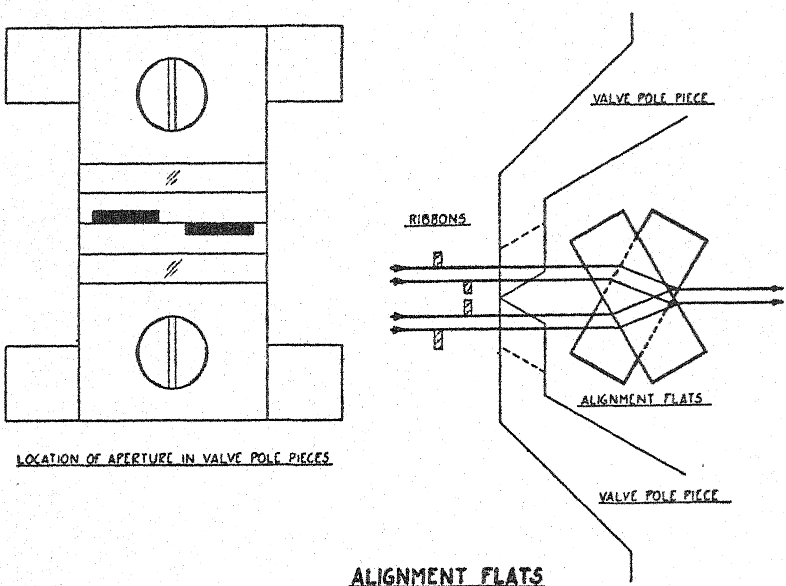
Uniform exposure, to give a uniform field across the width of the sound track, is obtained by adjusting the flat-ribbon filament lamp. These lamps, incidentally, must be carefully selected inasmuch as there is a wide variation in lamps from any one manufacturer.

If the two halves of the track are recorded in-phase, the output when reproduced on a balanced machine will be a measure of the cancellation.

In the variable area Class A push-pull system, the images of the triangular slits are so located as to be symmetrical with respect to the center of the slits, the center of the track being placed to correspond also with the center of the slits. The residual width of the track will then indicate a measure of the balance.

(b) Dimensions of Studio Push-pull Tracks

In the variable density Class A push-pull system, there are two



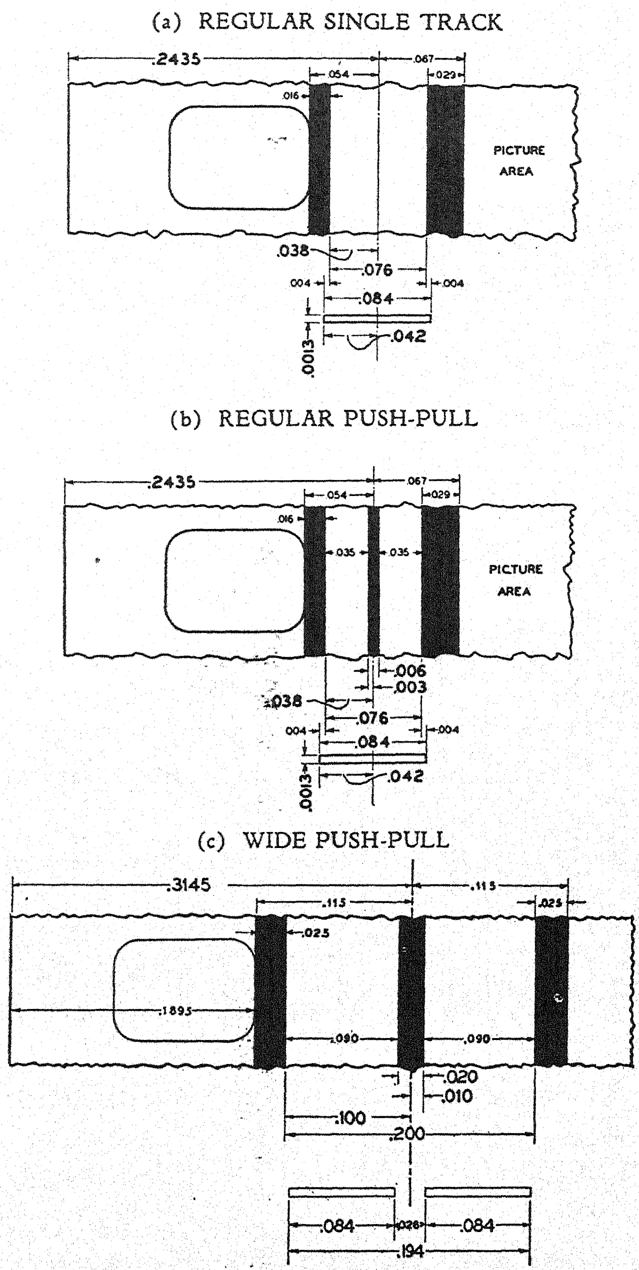
—*Courtesy Electrical Research Products, Inc.*

Figure 19 — Alignment flats of four-ribbon valve and associated optical system.

widths of sound track in current use in the studios: Regular (76 mils) and wide (200 mils). Regular push-pull occupies the same width as that of the original single track with the exception that there are two 35 mil tracks separated by a 6 mil septum, these tracks being scanned by the usual 84 mil slit. These dimensions are also used for push-pull movietone prints.

Wide push-pull tracks consist of two 90 mil tracks separated by a 20 mil septum. Each of these tracks is scanned by an 84 mil slit.

These three types of sound track, illustrated in Figure 20, have been adopted as Working Standards by the eight major studios. The dimensions shown in the larger sized numerals must be maintained as accurately as possible. Other values are nominal and for information only. The areas shown in black are controlled by matting the negative in the recorder—not by matting in the printing machine. This method, which limits the effective track width on the film itself, greatly reduces the effect of weaving in recorders, printers and reproducers. The volume reduction due to the slight decrease in track width in comparison to widths previously used, amounts to only a fraction of a decibel and is well justified by other advantages obtained.



All dimensions in inches.

Figure 20 — Working Standards for: (a) Regular Single Track, (b) Regular Push-pull Track, and (c) Wide Push-pull Track.

For studio use the wide push-pull track has several advantages Some of these are:

1. Greater inherent fixed noise reduction is obtained, thus reducing the amount of bias current necessary for quiet operation. This decrease in bias current also reduces the breathing effect.

2. Regular and wide push-pull tracks may be intercut on the same reel and projected by a push-pull type reproducer, adjusted for the wide track. This is accomplished by printing the regular track through a wide aperture and in the area occupied by one of the two wide push-pull half-tracks. When the regular and wide tracks are intercut the two will reproduce on the same system, since the regular track utilizes only one side of the push-pull photocell arrangement as the track in the path of the other scanning slit is black. Hence, with a single adjustment the one reproducer will play both types of track with a difference only in volume efficiency when a change takes place from one track to the other.

3. Failure of one pair of ribbons still provides a standard track in case of emergency.

4. The track is separated from the sprocket holes by 25 mils compared to 16 mils in the narrow track. This increased barrier provides more isolation from 96 cycle sprocket hole modulation.

The advantages of the regular push-pull track, compared to single track, are:

(1) The same type of reproducer can be used for original as well as movietone push-pull.

(2) A four-to-one optical system can be used, which materially reduces the effective slit width on the film as well as a reduction in intermodulation provided ribbon amplitudes are limited in the case of variable density.

(3) Greater inherent noise reduction as discussed above.

(c) Squeeze or Matted Track

As push-pull reproducing systems have been installed in only a small percentage of the theatres, new techniques have been applied to single tracks, in order to secure a greater amount of noise reduction by a method which allows the use of the same type reproducer as previously used.

In the single track variable density method of recording, noise reduction by means of bias has been increased to the maximum consistent with good quality. Since this has resulted in a volume range still considered inadequate, other means of obtaining increased noise reduction must be used. As volume range is a function of the signal to noise ratio, equivalent noise reduction may be obtained by decreasing the width of the track and simultaneously increasing the percentage of modulation where the required output is considerably less than maximum.

Figure 21 — Squeeze or matted track.

A track so reduced is commercially known as a "squeeze" or "matted track."

This technique can be applied with excellent advantage to very low-volume scenes which are exemplified by whispers or subdued conversational passages, or in the use of a very low musical background to silent scenes.

One method used to obtain matted tracks was the insertion of an inverted V-type mask in the optical path, usually in fixed steps which reduced the track width to 25% or 50% of the available width.

The use of the above V-type squeeze track requires that the slit of the reproducer be evenly illuminated in order that the increase in modulation and decrease in track width exactly compensate. If this illumination is not even, matted portions will be reproduced at different levels depending upon the degree of non-uniformity of the illumination.

To overcome this difficulty, a more recent procedure is to use a W-type mat which reduces the track width from the center as well as from both outside edges (see Figure 99), and so lessens the effect of non-uniform slit illumination.

This type equipment, which is schematically illustrated in Figure 22, also provides a continuously variable mat. The attenuator motor and the matting motor are driven by a foot controlled Selsyn motor sys-

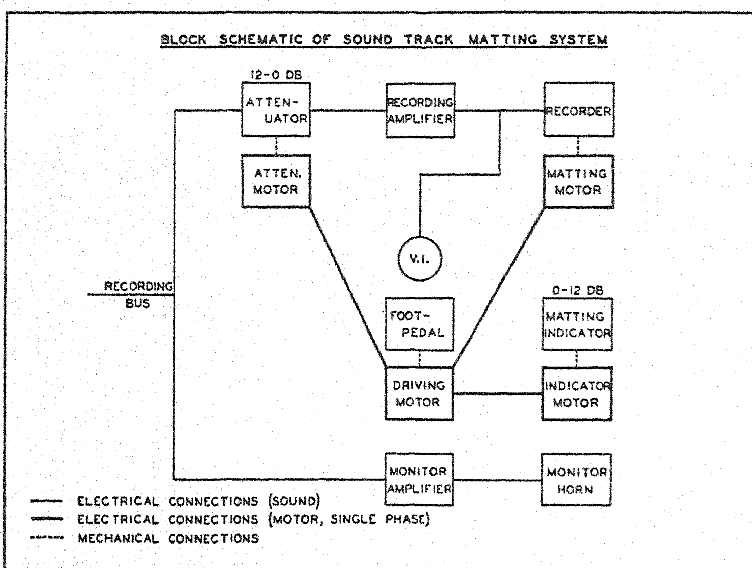


Figure 22 — Schematic diagram of equipment connections for squeeze track recording.

tem which provides the continuously variable mat, and simultaneously removes the proper amount of attenuation from the circuit to increase the modulation. The matting indicator (driven by the indicator motor through the driving motor), shows the operator the amount of matting applied, and it will be noticed that the amount of attenuation provided is exactly the inverse of the amount of matting available. From the volume indicator meter the operator is able to determine the amount of modulation upon the film and by proper control of the foot pedal, apply the necessary amount of matting which, through the simultaneously operated motor system, automatically provides the correct amount of attenuation.

Figure 21, an example of one type of "squeeze" or "matted track," shows in the upper portion of a sound track with no noise reduction applied. The center portion has a 6 db and the lower portion a 12 db noise reduction, a reduction in track width to 50% and 25%, respectively.

(d) Complementary Recording

Complementary recording provides a method, for use as an adjunct to present type noise reduction systems, to secure increased noise reduc-

tion and to partially suppress breathing effects. In addition to these benefits, inter-modulation between low and high frequencies of signals is reduced, and some overloading of the recording mechanism is avoided on abnormally high-volumed, low-frequency signals.

Modern methods of recording and reproduction do not permit much extension of the amount of noise reduction which can be secured in practice. This is due to several effects which accompany the use of biased noise reduction systems, such as breathing, distortion effects caused by the sluggishness of the bias component, and other items well known to the industry. For the method described below, these items, limiting the use of standard noise reduction systems, are partially overcome by distributing the signal load more uniformly upon the recording mechanism with respect to frequency. This is accomplished by pre-equalization and post-equalization methods of a type different than those heretofore employed for sound-on-film recording.

Figure 23 shows the circuits of two equalizers, one for use in the recording circuit to pre-equalize the recorded sound material, and the other for use with the corresponding reproducing channel to post-equalize this circuit in a complementary manner. When these two equalizers are so used, the amplitude-frequency characteristic of the reproduced sound material is unchanged because of the complementary characteristics of the two equalizers. Throughout the complete recording and reproducing channels, more than normal gain is needed to compensate for the equalizer losses, which are 14 db for the equalizers of Figure 23. Referring to the figure, it is noted that the recording equalizer has an insertion loss of 14 db at 100 c.p.s., 7 db at 1,000 c.p.s., and very little loss for the high frequencies. In general, most of the transition from 14 db loss to zero loss occurs in the frequency range from 300 to 3,000 c.p.s. The half-loss frequency of 1,000 c.p.s. is one of the design parameters of the equalizers.

A large part of the energy content of sound signals lies in the low frequency range from 200 to 500 c.p.s. Insertion of the above pre equalizer into a normal recording channel without any change in channel gain removes a large part of the load from the recording mechanism and the film, leaving a high-frequency content of approximately the same level. Due to the removal of the low-frequency load, it is now possible to increase the recording channel gain, thus securing a greater ratio of high-frequency signal to static surface noise; that is, an increase in noise reduction. Subsequent post-equalization does not destroy this increased noise reduction because of the concentration of surface noise in the upper part of the frequency spectrum. In addition to this increase

in noise reduction, a further increase is sometimes obtained due to the removal of restrictions applying to the noise reduction equipment as discussed below.

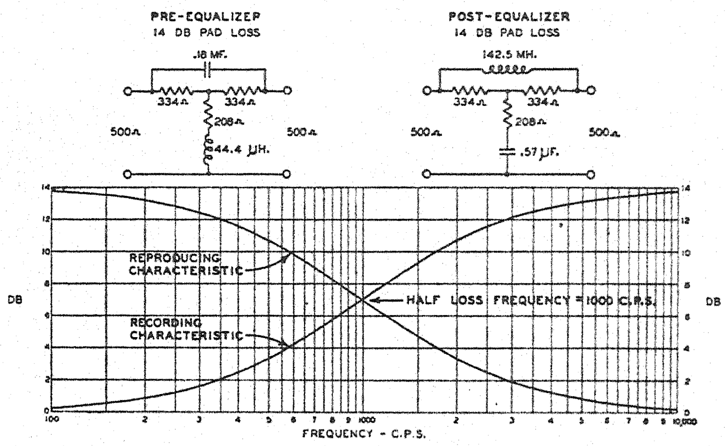


Figure 23 — Equalizer circuits used in complementary recording.

Breathing is caused by the action of the noise reduction system in placing the modulator in a position to handle the signal. Normal operation of noise reduction systems consists of a bias on the modulator or a reduction in track width, so that the minimum amount of static surface noise is present for the no-signal condition. Opening up of the recording mechanism or track width by the rectified bias signal increases the surface noise, causing the effect known as breathing. The greater the volume of the recorded signal, the greater the breathing effect.

As already mentioned, most of the signal energy lies in the low-frequency range which means that much of the breathing is produced by the lower frequency components of the signals. Hence, use of the above type of pre-equalizer greatly decreases breathing because the recording mechanism is not modulated nearly as much by the low-frequency content. Tests have shown that breathing is practically eliminated by this method even on piano and organ tracks. It may be mentioned that in normal recording, breathing caused by the high-frequency content of signals is not nearly as objectionable as that from low-frequency components because of the low-energy content and masking effects.

Some other beneficial effects derived from this type of pre-equalized recording are: Possibility of the use, in some cases, of a greater amount

of noise reduction because breathing, which normally tends to limit the amount of noise reduction that can be used, is reduced; the reduction of spurious signal products, arising from intermodulation, because of the lower level of the low-frequency components; the reduction of bias current components for the same reason; and the reduction in the amount of wave top clipping on steep wave front signals which is normally caused by the sluggishness or time lag of the bias signal. (See chapter on Noise Reduction.)

In conclusion, it may be mentioned that this method of recording accomplishes beneficial results for two basic reasons; first, because the energy distribution of acoustic signals lies primarily in the lower part of the audible frequency spectrum and, second, because film surface noise is concentrated in the upper part of the frequency spectrum. Reducing the level of the recorded signals in the low-frequency range does not materially increase the signal to static surface noise ratio in this same range, but on the contrary permits the over-riding of surface noise in the upper-frequency range. This, in connection with the reduction in breathing, decreased intermodulation effects, and the other items outlined above, constitutes the benefits which may be derived from this method of recording.

At the present time this method is in current use on original recordings, the post equalizer being used at the time of re-recording for standard release. It is hoped that within the very near future a sufficient number of theatres will be equipped with the post equalizer in order that the full benefits of the system may be obtained by release on the movietone print.

6. PHOTOGRAPHIC REQUIREMENTS OF VARIABLE AREA AND VARIABLE DENSITY SYSTEMS

The primary requirement of any optical recording and reproducing system is that the relative amount of light falling upon the photoelectric cell be, at each instant, proportional to the amount of light passed by the exposure device in recording.

This requirement is shared by both systems, but due to the difference in the exposing devices in the two cases, a different use of the film characteristics is employed, as covered in detail in the chapter on Film Processing.

Chapter IV

NOISE REDUCTION

By FRED ALBIN

Theoretically, an ideal electrical recording and reproducing system is one which will produce an electrical output similar in every respect to the electrical input (see Figure 24). Any deviation between the two is known as distortion, the two commonest forms of which are amplitude distortion and frequency distortion. Practically, how-

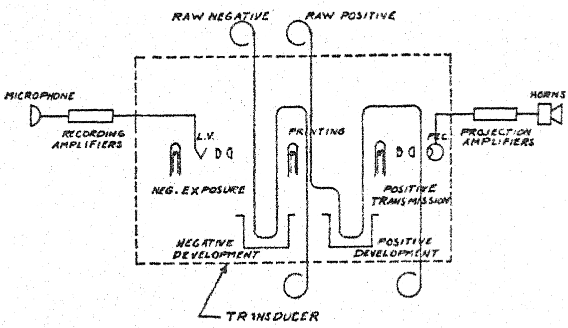


Figure 24 — Elements of sound recording and reproduction system.

ever, such a system will always present some distortion and will differ from the ideal in several respects, one of which is the introduction of "ground noise" during the recording process.

A measure of the efficacy of such a system is the volume latitude; the maximum limit of which depends upon the overload level of the system and the minimum limit of which depends upon the ground noise present on the record. Figure 25 illustrates this, showing the maximum and minimum levels and giving the volume latitude as the difference between these two levels. It is evident, therefore, that a reduction of the noise level increases the volume latitude and so enhances the value of the system.

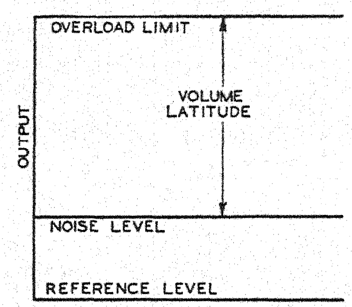


Figure 25—Diagram illustrating the volume latitude of reproducing systems.

This is further brought out by referring to Figures 92 and 96, where it can be seen that the difference in level between the two curves at any particular volume-value is the volume latitude of the system in question.

It will be noticed that the slope of the noise level curve and the minimum volume curve are equal, as any ground noise which is on the film will be amplified proportionately to the "wanted" signal contained on the film. Therefore, increasing the power of the installation will not increase the volume latitude unless the ground noise is held constant, or in other words, unless some system of noise reduction is used.

This chapter will be devoted to the manner in which noise reduction may be accomplished and to the methods of application to the variable area and variable density recording systems.

1. METHOD OF APPLYING NOISE REDUCTION TO VARIABLE AREA RECORDINGS

The variable area method will be discussed first, as the application of noise reduction is more easily visualized due to the type of track produced by this system. As explained in Chapter III, the first type of variable area record consisted of a track which had the appearance of an oscillograph record on which the area to one side of the oscillogram was transparent and the other side opaque. This is illustrated in Figure 26, where it can be seen that one side of the track is quite clear (except for

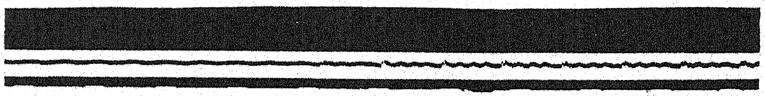


Figure 26-A — Oscillogram of a speech wave.

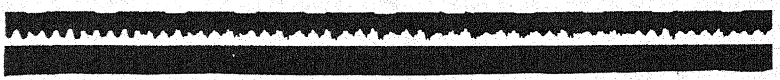


Figure 26-B — A single envelope variable area recording of the speech wave of Figure 26-A.

—*Courtesy RCA Manufacturing Co.*

fog grains of silver, dirt, abrasions, etc., present on and in the clear film, but which are not discernible in the cut). These foreign particles, in passing the scanning aperture, interrupt the light beam in the same manner as a modulation wave, and so produce undesirable noise in contrast to the signal itself, which is producing the wanted sound. Since these particles are located at random throughout the clear area of the track, their power of modulation is proportional to the width of the clear track area (Figure 27). Without modulation, that is, with no signal recorded on the track and without noise reduction, the widths of the

clear and opaque areas are equal. The division line between these two areas is known as the zero line or base line, and is the axis of the recorded wave during modulation. The width of the clear area determines the maximum undistorted amplitude of the modulation, that is, any

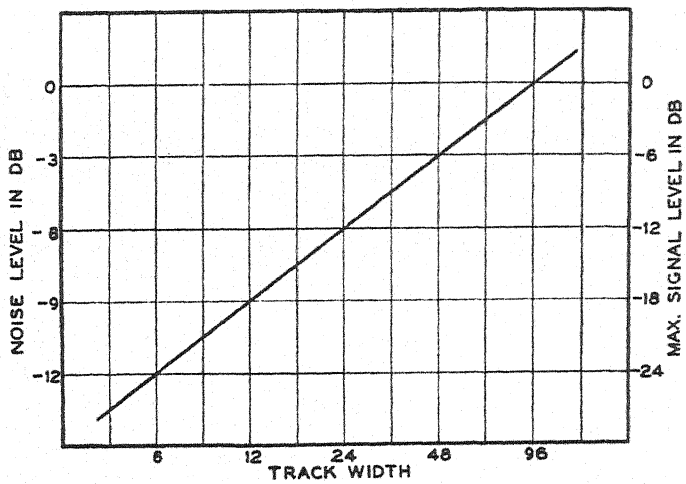


Figure 27 — Relation between track width and noise level and maximum signal level.

signal which has an amplitude greater than one-half of the track width will obviously extend beyond the track, and the peaks of the waves will be lost and harmonics introduced. At low modulation levels the clear area width is excessive and as a consequence the foreign particles mentioned above are present in a relatively large extent, which results in a serious prominence of ground noise. The obvious remedy is the reduction of the excessive width of the clear portion of the track during low modulation levels. Such a system, when applied to recording methods, is known as "noise reduction."

The first experimental plan for accomplishing noise reduction in the variable area system consisted of applying a direct-current bias to the modulator circuit. This direct-current bias caused the zero line to be shifted toward one side of the track, during no-signal periods, thus reducing the width of the clear area to some pre-determined value. During modulation periods, part of the signal energy was diverted from the modulator and used to reduce this bias current, thereby moving the zero line toward the center of the track to allow sufficient track to record the increased modulation.

However, mechanical difficulties were introduced by the weaving of the film when passing through the projector, as the modulation at low levels often moved outside the area of the scanning beam and part

of the wave was lost. Consequently, this system never was used commercially.

This difficulty was overcome by removing the bias current on the galvanometer and adopting a biased shutter which moved into the track area from the clear side of the film toward the zero line at low-modulation levels, leaving the same net width of clear track with the zero line

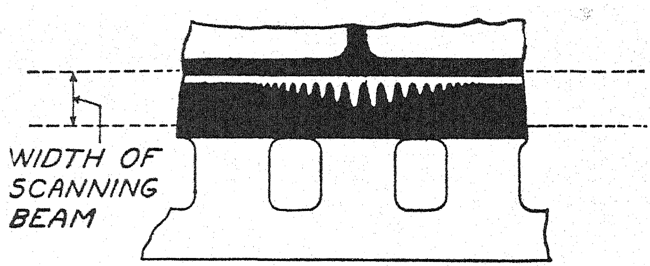


Figure 28-A — Variable area sound track with noise reduction accomplished by variable bias of galvanometer.

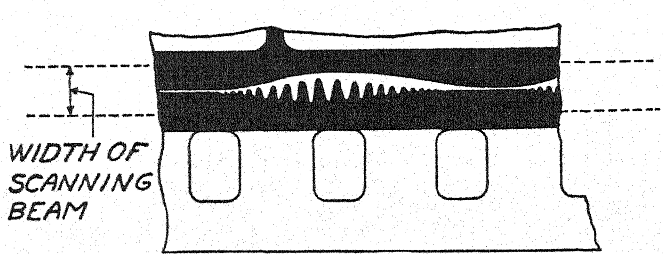


Figure 28-B — Variable area sound track with noise reduction accomplished by use of shutter.

—From the Journal of the S.M.P.E., August, 1931. "A Shutter for Use in Reduction of Ground Noise," E. W. Kellogg and C. N. Batsel.

still located in the center of the whole track, and avoiding the loss of modulation resulting from weave of the film. The bias current on the shutter is reduced, at high-modulation levels, by using part of the rectified signal energy to cancel this bias current. Figure 28-A illustrates an experimental sample recording made under the first plan, and Figure 28-B represents a sample recording made under the second plan. These single envelope variable area records are made with an aperture as shown in Chapter III.

Later, a different type of aperture, in the form of an isosceles triangle, was adopted to replace the original square aperture. An image of this triangle is moved in a vertical direction, and at right angles with respect to the scanning slit, as shown in Figure 29-A, resulting in a record as shown in Figure 29-B.

The image of the common area of the optical slit and triangle (shaded area in Figure 29-A) thus varies in horizontal length in accordance with the signal, and the track produced is known as a bilateral

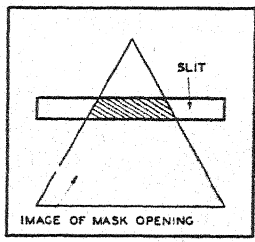


Figure 29-A — Type of aperture used in bilateral variable area recording.

—Courtesy RCA Manufacturing Co.

Figure 29-B — Sample of variable area bilateral recording.

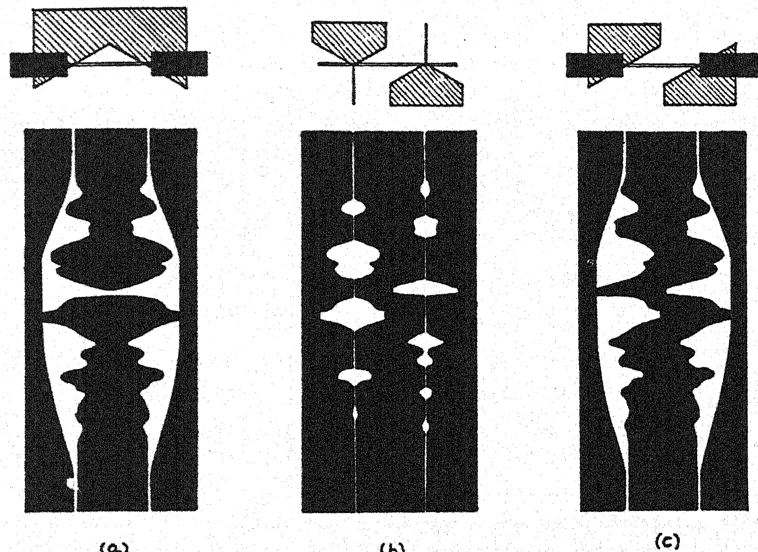
track, as it is a double envelope track identical in modulation on both sides of the zero line. The modulator may be biased to reduce the horizontal length of the image which results in a corresponding reduction of the area of the clear portion of the positive track. This method accomplishes noise reduction without the mechanical difficulties experienced with the original method.

From this followed the adoption of a double track system, which is produced by replacing the single triangular aperture with two opposing triangular apertures as shown in Figure 30-(b). An optical image of the two triangles is formed at the scanning slit after being reflected from the galvanometer mirror. The apexes of the triangular images fall on the center of the slit at the "no-signal" position, and are spaced apart half the length of the slit. The signal to be recorded causes the galvanometer to vibrate and to shift this image, according to the signal, in the direction shown. Thus, one triangular image will record the positive portion of the signal, and the other image the negative portion, which produces a track as shown in Figure 30-(b). This type track is known as "push-pull, Class B, variable area track."*

If the aperture is so designed that the mid-point of the altitude of both triangles falls on the center of the scanning slit at the "no-signal" value, Figure 30-(c), the resulting recording is known as "push-pull, Class A, variable area track."* This aperture was so modified that

*NOTE: As will be explained later in this chapter, Class B push-pull records are unsuitable for release print purposes at the present stage of advancement of reproducing equipment. For this reason, Class B push-pull recording, in both the variable area and variable density systems, is not used, and for purposes of release print nomenclature the terms "Class A" and "Class B" are dropped, and the term push-pull record always designates a push-pull Class A track.

the image falling on the scanning slit consisted of two opposing right triangles instead of the two opposing isosceles triangles, in order that noise reduction of the shutter type might be easily applied. The re-



—From the Journal of the S.M.P.E., September, 1937. "The RCA Recording System and Its Adaptation to Various Types of Sound Track," G. L. Dimmick.

Figure 30 — Types of variable area apertures with noise reduction shutters, with the resulting sound tracks.

sulting track is shown in Figure 30-(c), which is similar to the track of Figure 29-B, but with the two half-tracks 180° out-of-phase.

The most recently developed aperture and the one currently used in production, except where push-pull is employed, is shown in Figure 30-(a). This "M-shaped" aperture is a negative of the single track aperture of Figure 29-A and its use results in a track shown in Figure 30-(a). Noise reduction, that is, reduction in the width of the transparent portion of the track, is secured by the use of a double-shutter mask, each shutter biased toward the center of the track during periods of low modulation. Part of the signal energy is diverted from the modulator, rectified, and then used to cancel the biasing current on the shutters to allow greater track width at periods of high modulation, as illustrated in the center portion of Figure 30-(a).

In sound recording, the majority of signals are unsymmetrical about their base line,* having greater amplitudes during the compression part

* While this condition appears to be true, there is considerable difference of opinion among authorities on the subject.

of the wave. For this reason the galvanometer should be so phased with the sound wave that the compression part of the wave is recorded toward the center line, and away from the shutter side of the track. This provides a greater safety margin for high-peak waves as the total amount of available track on a bilateral record may be instantly utilized by the compression part of the signal. However, on both outside quarters of the track, the shutter, during periods in which the noise reduction is operating, masks part of the track, and high amplitude waves are apt to be cut off during the time the shutters are being removed from the sound track area. This timing action of the shutters is described in detail below.

2. METHOD OF APPLYING NOISE REDUCTION TO VARIABLE DENSITY RECORDINGS

The variable density recording system employs as a light modulator the light valve, which has a normal spacing between the two ribbons of one mil. Noise reduction may be applied to such a system by reducing the mean spacing of the ribbon to some predetermined value, resulting in a reduction of negative exposure with a consequent reduction of positive transmission. This has an effect similar to reducing the clear portion of the track of the variable area record. Part of the incoming signal is then diverted through an appropriate network to act in such a way as to increase this mean ribbon spacing as the modulation of the signal increases. This increase of mean spacing is sufficient to accommodate the increased input.

More recent designs of variable density recorders use a four-ribbon valve to expose two tracks in the area formerly occupied by a single track. The exposure of one track is controlled by one pair of ribbons and of the other track by the other pair. The modulation of each pair of ribbons may be in-phase or out-of-phase by 180°, depending upon the external connection. An in-phase relation results in a track which is the same as that produced by a two-ribbon valve with the exception that a septum divides the sound track into two parts. An external connection resulting in an out-of-phase condition produces a push-pull record.

This push-pull record may be either Class A or Class B. In the Class A type each pair of ribbons receives one-half the total signal energy, and has a fixed spacing of over one mil. The resulting track consists of two parts, one part being identical with the other, yet with the two parts 180° out-of-phase. A sample Class A push-pull, variable density recording of low frequency is shown in Figure 31-A.

In the Class B type of recording the ribbons are still connected 180° out-of-phase, but in this case one pair of ribbons receives the positive half and the other pair receives the negative half of the signal energy. This results in a type of track which is very similar in appearance to the Class A push-pull record. A sample of Class B push-pull variable density recording is shown in Figure 31-B.

While it will be shown later that the Class B system of push-pull recording is ideal from a standpoint of maximum noise reduction, it

—*Courtesy Electrical Research Products, Inc.*

Figure 31-A — Sample of push-pull, Class A, variable density recording.

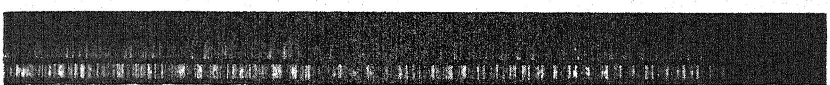


Figure 31-B — Sample of push-pull, Class B, variable density recording.

NOTE: Figure 31-A is the usual 100 mil sound track used on release prints (actual size), while Figure 31-B is a 200 mil sound track frequently used in studio work (actual size). The 200 mil track is of course reduced to 100 mil track for release.

has, nevertheless, been superseded entirely by the Class A system because of practical difficulties. Reproduction of Class B recording requires very careful balance between the positive and negative halves of the cycles in both an amplitude and phase relation if low distortion requirements are to be fulfilled. When the recording is reproduced, the relative sensitivities of the two photo-electric cells and their associated optical and electrical circuits must be maintained exactly in balance for all frequencies within the recording range.

In practice it has been found difficult to maintain this necessary balance and the fidelity sacrificed was not offset by the improvements realized in noise reduction.

The problems encountered in the variable density system, due to the film characteristics employed by this system, add still further to the distortion by the Class B type recording. This can be seen by reference to Figure 36, where it should be noted that the curve below the

value of T_1 becomes non-linear and as this section of the curve is utilized by the Class B recording, distortion results.

3. DEFINITION OF TERMS

Before taking up the effect of noise reduction on the reproduced noise level from both variable area and variable density records, it will be necessary to establish the definition of certain terms.

In order to explain, in common terminology, the operation of noise reduction as applied to both the variable area and variable density systems, a nomenclature has been chosen, which, while not elucidative to either particular system, is descriptive when applied to both systems.

The total track width, which is the area of the film scanned by one projection scanning beam, will be termed "track." The "carrier" is the medium which, when varied in magnitude, constitutes recording. This is not to be confused with a popular specific type of noise reduction control circuit known as the "Carrier Noise Reduction System," described later. The carrier amplitude in the variable area system will be defined as the width of the clear portion of the track and in the variable density system as the mean positive projected transmission. The percentage modulation, that is, the percentage of the track covered by modulation, is the ratio of the modulation amplitude to the carrier amplitude expressed in per cent. Margin, which is the excess of the opening of the valve over the necessary amount to carry the signal (variable density), or the amount of clear track between the peak amplitude and the modulation and that portion of the track masked by the noise reduction apparatus (variable area) is expressed mathematically as

$$\text{Margin} = 20 \log \left(\frac{100}{\text{percentage modulation}} \right) \text{db} \qquad (8)$$

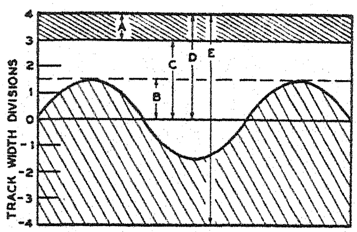
$$= 20 \log \left(\frac{\text{carrier amplitude}}{\text{modulation amplitude}} \right) \text{db} \qquad (9)$$

For example, if the carrier amplitude in the variable density system is 25% and the modulation amplitude is 12½%, the percentage modulation is 50% and the margin is (20 log 2) 6 db.

The physical meaning of these terms is shown in Figure 32.

In practice, some value of margin in the order of 6 db is used for several reasons: First, there is a time delay in the operation of the noise reduction equipment and the margin assists the bias cancellation in its

speed, that is, it allows the noise reduction equipment a longer period of time to clear the way for a signal of increasing modulation; second, the adjustment of margin is usually made with a sine wave, which does not have as large a peak factor as do wave forms encountered in ordinary



A = Amount of noise reduction.
B = Modulation amplitude.
C = Carrier amplitude.
D = Clear portion of track with no signal and no noise reduction.
E = Track.

$$\text{Percentage Modulation} = 100\frac{B}{C} = 100\frac{1.5}{3} = 50\%$$

$$\text{Margin} = 20\log\frac{100}{50} = 20\log 2 = 6\text{ db}$$

Figure 32.

recording, and since the response of the noise reduction control circuits is usually approximately proportional to the effective value of the input level, its response when predetermined by the sine wave would be insufficient to accommodate the higher modulation of a complex wave.

4. DISTRIBUTION OF NOISE IN THE RECORDING BAND

Investigations of the different sources and causes of noise have been made and the results indicate that if sufficient care is taken in the processing of the film, the greatest contribution to noise is the irregularity of the grain grouping of the photographic emulsion, as well as the irregularity in transmission characteristic of the gelatin and the base of the film itself. Again, after the film has been in use for some time it accumulates various kinds of dirt, becomes scratched, and otherwise depreciates from an ideal medium for sound recording.

Figure 33-A gives the distribution of the noise level over the frequency range used in recording, while Curve B, Figure 33-B, gives the *effective* noise level due to the non-uniform intensity curve (Curve A) of the ear. It must be noted that there is actually more energy in the low

frequencies than in the high, but the sensation level curve of the ear (Curve A) indicates that frequencies in the mid-range are actually more disturbing.

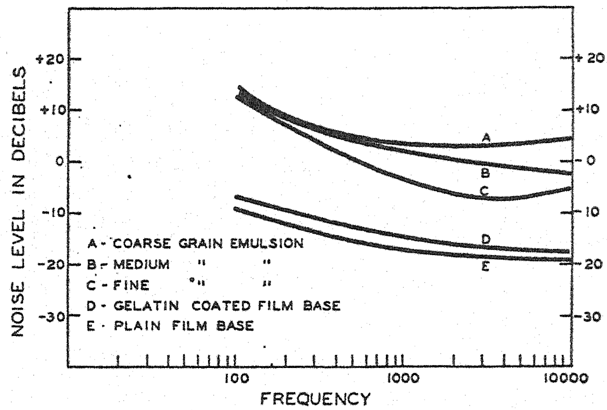


Figure 33-A — Distribution of noise level over the frequency band used in recording.

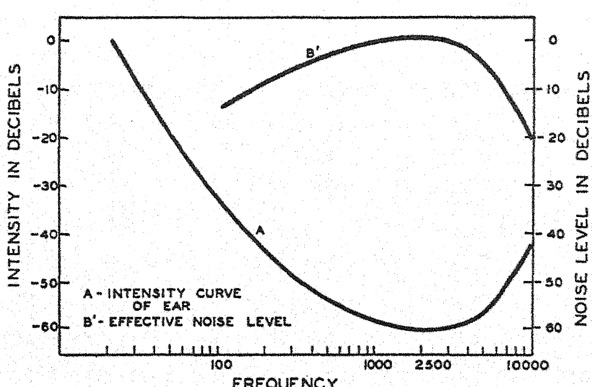


Figure 33-B — Curve A shows the sensitivity of the ear as a function of the frequency, while Curve B′ shows the effective intensity of noise produced by a film record over the frequency range used.

5. "CARRIER REDUCTION" METHOD OF NOISE REDUCTION

As previously mentioned, the most commonly used method of noise reduction is the "carrier reduction" method, commonly known as either the "bias" or "shutter" method depending upon the choice

made, wherein the carrier amplitude is reduced to a value which will accommodate the modulation amplitude with only a small margin for tolerance. Such a system is illustrated in Figure 34.

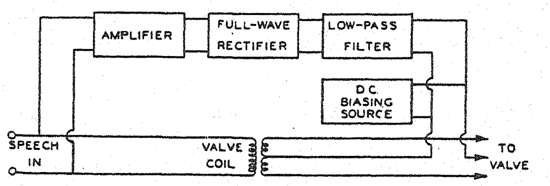


Figure 34 — Block schematic of a conventional noise reduction system.

The reproduced noise level is usually expressed in terms of amplitude and from a variable area record the reduction of amplitude is proportional to the square root of the reduction of the clear track width.

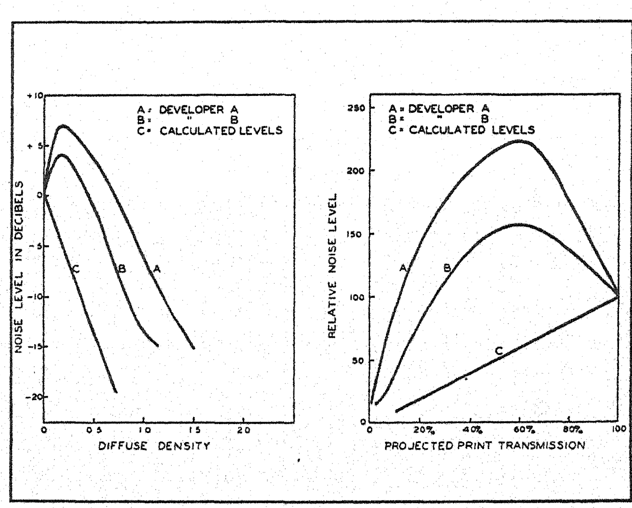


Figure 35-A — Curves showing the theoretical and measured relations between noise level and density of positive film developed in different developing solutions.

Figure 35-B — Curve showing the theoretical and measured relations between the relative noise level and the projected print transmission of positive film developed in different solutions.

For example, if a reduction of the noise level of 4 to 1 is desired, it is necessary to reduce the width of the clear portion of the track to 1/16 its initial value.

From variable density records the noise level is also proportional to the reduction of power, but in this case the amplitude of the ground noise level bears a close relation to the transmission of the film over a wide range of density, as shown in Figures 35-A and 35-B. For example, if a reduction of noise level of 4 to 1 is desired, the mean positive projection transmission must be reduced in the same ratio. This transmission is proportional to the valve spacing and consequently the desired noise reduction is accomplished by reducing the valve spacing by the desired ratio. (See Appendix, Page 527.)

In both the above cases the noise reduction is expressed in decibels as 20 log (reduction of amplitude of noise level) = 12 db.

This may be illustrated by considering the alternating-current analogy. The foreign particles present in the clear portion of the track may be considered as distributed more or less uniformly and these contribute to the reproduced noise level in a power ratio according to the amount of clear track projected. In the electrical case, if two equal alternating currents which have the same frequency and which are in-phase are impressed upon a circuit, the resultant current is equal to the sum of the two. The power in the circuit, however, is proportional to the square of the current and therefore the power has been increased four-fold by doubling the current. This condition is similar to noise reduction in variable density as any change in the transmission affects all noise sources simultaneously and the reduction of noise is proportional to the reduction of light. Therefore, the reduction of noise level is proportional to the reduction of mean positive projected transmission.

However, if the two currents mentioned above are not in-phase or are of a different frequency although of equal amplitude, the resultant current does not have the same linear relation to its components. In this case, the resultant current is equal to the square root of the sum of the squares of the component currents, or, if of equal value, to the square root of 2 times the value of either component current. Thus, the power, when the current is doubled, is also doubled. This is similar to the variable area case and shows that the reduction in noise level is proportional to the square root of the reduction of clear track width.

Therefore, to secure a 12 db reduction in the noise level, it is necessary to reduce the positive projected transmission in variable density recording to one-fourth its original value, while in the variable area system it is necessary to reduce the clear portion of the track to one-sixteenth its original value.

The following is an example of the application of this amount of noise reduction to both systems of recording:

Let it be required to reduce the noise level to one-fourth its value when noise reduction is not used. Then if the subscript "1" designates the unreduced noise level and the subscript "2" designates the noise level at one-fourth the original value:

$$NL_1 = 4\,NL_2 \qquad \text{where } NL = \text{noise level amplitude}$$

Variable Area

$$NL = K\sqrt{\text{clear track width}}$$

$$NL_1 = K\sqrt{CTW_1}$$

$$NL_2 = K\sqrt{CTW_2}$$

$$\text{but} \qquad \frac{NL_1}{NL_2} = 4 = \frac{\sqrt{CTW_1}}{\sqrt{CTW_2}}$$

$$\text{and} \qquad \frac{CTW_1}{CTW_2} = 16$$

$$CTW_2 = \frac{1}{16}\,CTW_1$$

Variable Density

$$NL = K\,(\text{positive projected transmission})$$

$$NL_1 = K\,(PPT_1)$$

$$NL_2 = K\,(PPT_2)$$

$$\text{but} \qquad \frac{NL_1}{NL_2} = 4 = \frac{PPT}{PPT}$$

$$\text{and} \qquad \frac{PPT_1}{PPT_2} = 4$$

$$PPT_2 = \tfrac{1}{4}\,PPT_1$$

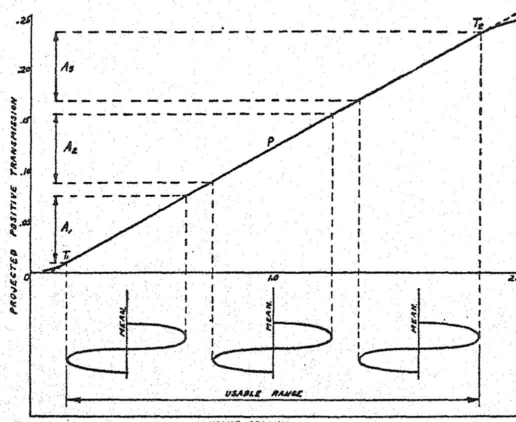
In both cases the amount of noise reduction expressed in decibels is the same, namely:

$$NR_{db} = 20 \log \left(\frac{NL_2}{NL_1}\right)$$

$$= 20 \log 4 = 12 \text{ db}$$

Returning again to the variable density system and considering the combined steps of film processing as a whole, the positive projected transmission is proportional to the valve spacing. The signal input, which controls the variation from a predetermined mean of valve spacing, therefore causes a resulting change in the positive projected transmission, provided the relations given above are kept linear. The design of the photo-electric cell circuit serves to convert the change of transmission to an alternating electrical current whose amplitude is proportional to the change in transmission. The mean transmission, therefore, has no effect upon the output signal level which is determined solely by the above alternating current—but the mean transmission represents the source of, and determines the value of, the ground noise.

The minimum ground noise level is therefore reached when the mean transmission is reduced to the lowest allowable value. The lowest allowable value is limited by the straight line portion of the curve, which gives the relation between negative exposure and positive projected transmission. The mean exposure must be greater than this lower limit by an amount equal to the amplitude of the modulating wave, and so the minimum mean exposure is directly dependent upon the modulation level. In this respect the variable density and variable area systems of recording are similar. As shown above, the reproduced signal level, being proportional only to the modulation amplitude or change of transmission, is independent of the value of the mean transmission and, as a consequence, a reduction in this mean transmission, provided it is kept above the minimum value determined by the modulation amplitude, leads to a reduction in noise level at a constant output level.



—*Courtesy Electrical Research Products, Inc.*

Figure 36 — Curve showing the relation between positive projected transmission and the ribbon spacing of the recording valve.

Figure 36 illustrates the linear relationship between negative exposure and positive projected transmission. A constant amplitude of modulation is shown which is variable about three different mean values—it is evident that the output level in these three cases is the same.

To avoid amplitude distortion at any frequency, the output level of the recording system must be proportional to the exposure modulation amplitude. It is commonly known that the output level may be varied by changing the print transmission and that the output level is proportional to the percentage of modulation. When the print transmission is varied as a consequence of bias on the valve, the percentage modulation varies in an inverse manner, but since the output level is

proportional to the product of these two factors and as they change in an equal but inverse manner, the output level is unaffected.

When the illumination in the printing process is varied, the positive mean transmission and the positive change of transmission due to modulation are both varied, and in the same proportion. Therefore, the signal and noise levels vary together. This effect is used as a means of signal adjustment, but not as noise reduction.

However, for the above conditions to be true the exposure illumination and development conditions of both negative and positive films must be held constant. Variation of mean transmission as a result of varying illumination and development will definitely affect the reproduced level as well as the mean transmission, and as a consequence it cannot be used as a method of noise reduction.

6. "SPLIT-CHANNEL" METHOD OF NOISE REDUCTION

Another method of noise reduction illustrated in Figure 37 makes use of the frequency distribution of the film noise in the recording band. Referring again to Curve B of Figure 33-B, it is apparent that the higher frequencies represent greater effective noise level. The noise level may be reduced by the use, in the reproducer circuit, of an equalizer which causes an attenuation of these higher frequencies, but this leads to a corresponding loss of the high frequencies of the recorded sound as well as a reduction of the noise level.

There are two methods in use employing this principle of noise reduction.

In the first method, where only a single track is used, the high frequencies of the recorded sound are pre-equalized by an amount equal to the loss introduced by the post-equalizer, resulting, for all practical purposes, in the reproduction of a signal without frequency discrimination, but with a reduced noise level. (See Chapter III, Complementary Recording.)

The second system requires the use of two tracks, one for the low frequencies and the other for the high frequencies. The frequency band of the recorded sound is separated into two branches of the circuit by appropriate filters (see Figure 37), each branch operating a valve which utilizes only half the sound track. Thus the low frequencies are recorded on one-half the track and the highs on the other half. When this track is projected, a low-pass filter is inserted in the circuit from the low-frequency half of the track, effectively eliminating all noise above the cut-off frequency of the filter, and the high frequencies are reproduced

from the other half of the track in the conventional manner. This output is recombined with the output from the low-frequency side and

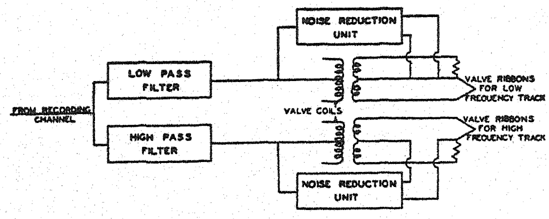


Figure 37 — Block schematic of two-channel high speed bias circuit.

fed into the same projection amplifier system. In recording this track, noise reduction of the carrier-reduction type is used only in the high-frequency circuit.

Due to the fact that the low frequencies have been eliminated from this branch, the time of operation of the noise reduction in this circuit may be accelerated, thus reducing distortion due to valve clash on the initial wave front, which is caused by too slow a removal of the valve bias when a signal of high modulation is introduced. The high frequencies in this branch of the circuit have a masking effect on the noise, which further reduces the noise level.

7. "AUTOMATIC ATTENUATION" METHOD OF NOISE REDUCTION

Another system of noise reduction which has been used with a certain degree of success in some instances, consists essentially of an automatic attenuator, sometimes known as a threshold limiting device, placed in the projection circuit to attenuate the transmission during periods in which no modulation exists. A control circuit, operated by the same energy which produces the output, actuates the attenuator and restores transmission of speech energy to a normal value during modulation periods, at which time the presence of modulation is depended upon to mask out the ground noise. This method has the added advantage of reducing noise from other sources than the film, such as extraneous noises picked up by the microphone, etc.

8. NOISE REDUCTION CIRCUITS

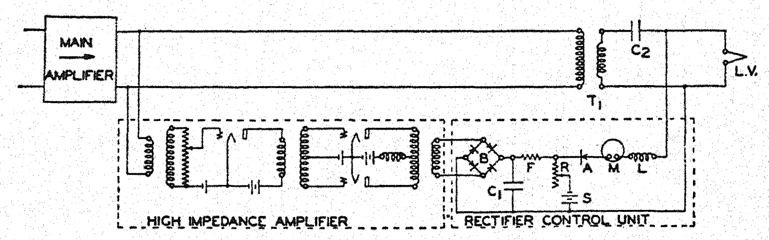
Figure 34 shows the noise reduction control apparatus for the common method of reducing noise by the reduction of carrier amplitude, accomplished either by the use of shutters or by biasing the modu-

lator. The ideal case of reduction of noise by the bias method, in which the bias follows exactly the envelope of the wave, is shown in Figure 30-(b). Any bias method would operate in this fashion if its response were in direct proportion to the input for both amplitude and phase. For this to be so, all amplifiers, rectifiers, etc., would have to have exactly linear amplitude characteristics, and all filters and other delay circuits would have to be removed from the system. This would lead to effectively zero instantaneous margin which would give the ultimate reduction in noise by the carrier reduction method.

However, in practical applications, a margin is necessary as the conditions outlined above are not fulfilled and the noise reduction control apparatus must be given sufficient time to operate and to remove the valve bias or shutter, as the case may be, when a high level of modulation requires a greater mean transmission or track width.

The noise reduction control apparatus consists of an amplifier circuit bridged on a special circuit which feeds the valve, the amplifier being coupled to a rectifier unit (see Chapter XXIII) which contains a filter. The output uni-directional potential from the filter is proportional to the amplitude (the value of which will be discussed later) of the speech energy and is used to restore the track width by removing the shutter or cancelling the bias.

Figure 38 is a schematic of a typical noise reduction system. The characteristics of this equipment are such that the direct-current output



—From the Journal of the S.M.P.E., May, 1932. "Western Electric Noiseless Recording," H. C. Silent and J. G. Frayne.

Figure 38 — Schematic diagram of circuits of noise reduction amplifier and control unit.

is proportional to the average value of the wave of speech input energy, and when this output is used to cancel the bias current, the net bias current is inversely proportional to the speech level.

The sensitivity of the circuit is regulated by adjusting the gain of the noise reduction amplifier and is so adjusted that when the modulator amplitude is maximum the bias current is completely cancelled. This adjustment is the margin adjustment. The rectifier prevents a reversal

of the bias current through the modulator, as it stops any further change of current when the bias current has been completely cancelled, regardless of any increase in signal level.

The modulator bias current is supplied by a battery in series with a variable resistance, this resistance being high with respect to the modulator resistance, and causing only a small bridging loss. This bias current is adjusted to the desired amount by means of the variable resistance, as indicated by the meter, this adjustment determining the amount of noise reduction applied.

Figure 39 illustrates the relation between current density and the resistance of a copper oxide rectifier. When the rectified signal voltage

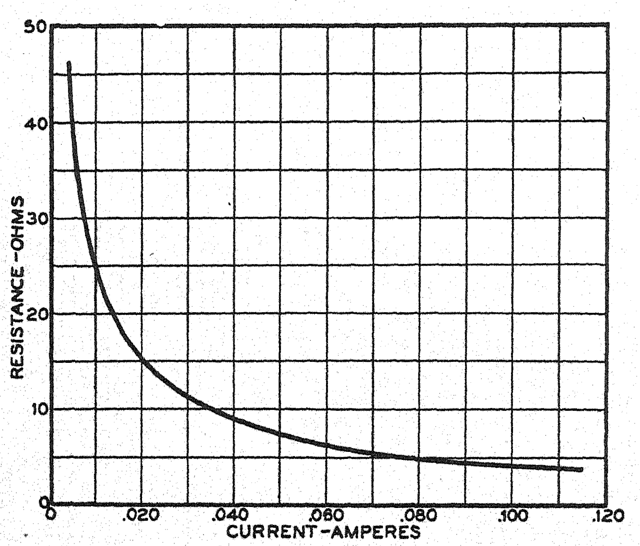


Figure 39 — Curve showing the relation between the resistance variation with current of a copper oxide rectifier.

is high, the resistance of this anti-reversing rectifier rises to a relatively high value and thus blocks any reverse valve bias current which would otherwise tend to flow. This signal rectifier has a slightly non-linear characteristic which is evident near the cut-off point.

The output of the rectifier is a uni-directional current which contains the double frequency of the original speech energy. If this component is not filtered out, it will modulate the valve, which in turn will record the same modulation on the film. Precautions must therefore be taken to prevent this modulation from entering the audible frequency range. Thus a limit to the speed of operation of the noise reduction

apparatus is established. This limit is established by that frequency which is the highest allowable below the cut-off of the system as established by the horns, filters, etc., in the circuit.

As an example, if the system will reproduce, as the lowest frequency, 20 c.p.s., and if 100% modulation is necessary for audibility at this frequency, then the maximum allowable rate of change of bias current will be approximately 1/80 or 0.0125 seconds. The necessary filtering circuit which follows the rectifier introduces a time delay in the circuit and should thus be designed for a cut-off frequency of about 20 c.p.s. as a maximum.

Due to the type of circuit and the fact that the full-wave rectifier impedance is constant, the periods for bias cancellation currents to build up and to decay are equal.

Another type of noise reduction circuit utilizes a vacuum tube rectifier in place of the copper oxide type. The filter which follows the rectifier consists of a simple condenser resistance network. The time constant of this circuit is fixed at approximately the same value as the previously described circuit.

9. TIMING CIRCUITS OF NOISE REDUCTION EQUIPMENT

The time constants of noise reduction circuits are determined in the following manner.

Consider Figure 40, which is a simplified circuit of the timing circuit of the noise reduction systems. In Figure 40

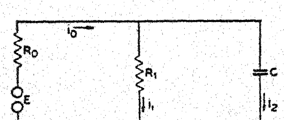


Figure 40 — Simplified circuit diagram of timing circuit of a noise reduction system.

$E =$ constant impressed voltage
$R_0 =$ resistance of generating source
$C =$ capacitance of condenser in farads
$i_0 =$ instantaneous current in R_0
$i_1 =$ instantaneous current in R_1
$i_2 =$ instantaneous current in C
$t =$ time

From the e.m.f. and current laws the following equations are obtained:

$$i_2 = \frac{dq}{dt} \tag{10}$$

$$i_1 = \frac{E - R_0 i_0}{R_1} \tag{11}$$

$$i_0 = i_1 + i_2 \tag{12}$$

From these equations may be derived an expression for i_1 in terms of t and the constants of the circuit:

$$i_1 = \frac{E}{R_1 + R_0}\left(1 - \epsilon^{-\frac{t}{RC}}\right) \tag{13}$$

Where $$R = \frac{R_0 R_1}{R_1 + R_0}$$

When $t = \infty$ then $i_1 = \dfrac{E}{R_1 + R_0}$

which is the maximum value of i_1. This is, of course, the theoretical maximum value, but practically, as the value of $\left(\frac{t}{RC}\right)$ becomes large at a very short time—as we normally measure time—after the switching operation, the value of t usually given for noise reduction systems is that value when $i_1 = 0.9$ (maximum value). (14)

This value of t is the operating time of the circuit.

When $$i_1 = 0.9\frac{E}{R_0 + R_1} \tag{15}$$

then $$t = 2.3\ RC \tag{16}$$

With this type equipment the bias current is the last stage amplifier plate current, which is reduced by the rectified signal applied to this stage as a grid bias voltage. To avoid "over-shooting" due to excessive signal, the circuit is adjusted so that this plate current approaches the cut-off point, beyond which the amplifier stage becomes non-linear. Use is made of this fact as it enables the equipment to provide a large margin at low levels and a small margin at high levels, which decreases to zero after the cut-off point has been reached and the modulation is 100%.

The response curve, that is, the relation between bias current and relative signal input, is linear over almost the total range used. However, at the extremes of the curve, that is, when the bias current is reduced to zero or when the signal reduces to zero, the relation falls off slightly.

Another noise reduction control system of the carrier-frequency type has its bias current supplied by the rectified output of a 20,000 cycle oscillator whose level is modulated by the rectified speech energy. The 20,000 cycle circuit is the "carrier" for this particular equipment, consequently the name carrier-frequency. The output level is maximum when the signal level is zero. The response characteristics are to a

large degree controlled by the adjustment of the exciter voltage, and the bias and modulation level into the modulating tube.

Noise reduction control equipment used with a light valve requires an equalizer circuit in order that the response with frequency be made to follow the light valve resonant characteristic, which condition provides a constant margin at all frequency conditions at a given level.

The usual noise reduction systems, that is, those with a linear response between the speech input into the noise reduction system and

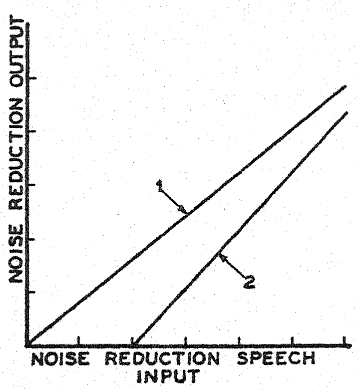


Figure 41-A — Curves showing the response characteristic of "linear response" (Curve 1), and "constant margin" (Curve 2) noise reduction systems.

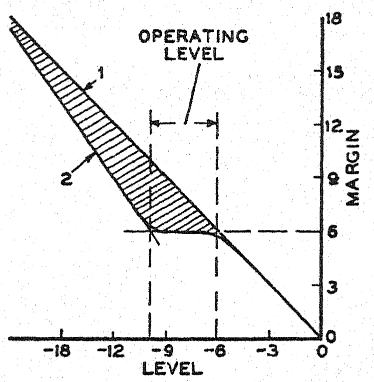


Figure 41-B — Curves showing the relation between margin and modulation level of "linear response" and "constant margin" types of noise reduction systems.

the output of the noise reduction system as described above, and as illustrated by Curve 1 of Figures 41-A and 41-B, provide too much margin in the operating region of the noise reduction system.

Another system, called the "constant margin" system, has been designed. This has a response characteristic as shown by Curve 2 of Figure 41-A. Figure 41-B illustrates the margin such a system provides at different levels, where zero level is a condition of 100% modulation. Curve 1 shows the relation of a linear noise reduction system. Curve 2 shows the margin provided at different levels by a constant margin system designed to operate at a margin of 6 db. As the modulation level is increased from 50 toward 100%, the margin approaches zero, which is the valve overload level. However, as the speech level decreases from 6 db, to say 10 db, the margin remains constant at a value of 6 db. The 10 db point (chosen arbitrarily for this drawing) is determined by the no-signal spacing of the valve ribbons and for any reduction in level below this point the margin increases in a linear manner as the modulation decreases.

Such a system further decreases the noise level for certain values of modulation. This further decrease in noise level is represented by the shaded area in Figure 41-B and represents an improvement over the linear-response noise reduction systems.

The noise reduction systems described above and those given in the chapter on recording methods, still leave a great deal to be desired from a consideration of the results obtained. In all probability a new method or a combination of old methods will eventually be worked out and used—a method which at the present time is not even anticipated.

Chapter V

RE-RECORDING AND PREPARATION FOR RELEASE

By KENNETH LAMBERT

The preparation of a picture for release involves processes which are not all of a technical nature. The philosophies of these processes, however, are tremendously important not only to the mechanical tools and operations used, but also to the very planning, writing and directing of the picture. From a strictly sound standpoint the interpretation given these philosophies may decide the choice of a recording system, the technique for using it, and the training and organization of the operating personnel.

Releasing a motion picture is about like preparing for a dinner party. You wish it to be one your friends will remember pleasantly. Many things will contribute to its success or failure. You hope your guests will come with keen expectation and will not be disappointed. Your home is always delightful, you are a good host, but you know that these particular people especially appreciate tasty, well prepared foods. Your favorite roast, its gravy flavored just to a king's taste; your particularly incomparable biscuits; and a relaxing evening to enjoy it all. Everything else is sure to be right but what if the cook fails you just this once! What if she is distracted by the children and puts too much salt in the gravy, or lets the biscuits burn? Well, at least she usually succeeds in making even the toughest meat moderately tender. You have cautioned the one who serves to be sure everything comes to the table at just the right temperature. At the last moment you go to the kitchen for one last smell, and then hope it will all arrive safely.

The producer has planned a perfect menu. His director has secured the finest foods. He has even grown most of them himself. The editor and recorder cook the meal, it is put into dishes by the laboratory, and is given to a theatre manager to serve.

It is difficult to consider the functions of editing and re-recording separately. Editing or "cutting" has been commonly considered to mean the assembly of the picture scenes and their corresponding sound in such fashion that the dramatic situations are presented forcefully in accordance with the plans of the producer. Re-recording or "dubbing"

was once the unavoidable process by which occasional necessary music or other effects were inserted into a sequence. This is no longer the case. Both the editor and the re-recorder must constantly have in mind the one common aim—how will the picture tell the story in the theatre? The editor must be more than picture-conscious, just as the re-recorder must be more than sound-conscious. Each must be a master of technique of various kinds, for motion pictures are inherently a technical product. Primarily, though, these men must be master judges of how best to put the producer's story across in a picture theatre. Their responsibility in this may exceed even the producer's, for they may be more familiar with the changing technical aspects of such presentation. They are showmen, and as such each is interested in the whole show, and not merely parts of it.

What makes up this show in a theatre? From a story standpoint, the sequences must be planned, shot and assembled into film with dramatic effectiveness, and this film must then be projected properly in the theatre. The picture must be focused sharply, with uniform, adequate illumination, on a carefully matted screen, of the most effective size for the particular theatre. The sound must be of the proper loudness, and free from distortions which might make it unintelligible, harsh, or disagreeable. The matter of effective presentation in the theatre now rests with the theatre owner and his manager and projectionist. Each must take a real interest in the mechanical performance of the theatre and all its facilities. The technique of good theatre presentation is discussed elsewhere in this book.

The process of releasing a picture divides naturally between several groups of people. The function of the laboratory is to intercut the negatives of the various picture and sound scenes in accordance with the "working" or "master" print prepared by the editor and re-recorder under the producer's supervision, and then to make the necessary release prints for use in the theatres.

The division of work between the editor and re-recorder in preparing the master print is dictated by convenience and economy. Their work is not entirely simultaneous and involves different techniques which are quite specialized but still of a nature permitting effective teamwork. Because of this, one editor can be "cutting" one picture while a re-recorder is working on the sound assembly of another picture. In this way, the equipment and personnel necessary for re-recording can be utilized most efficiently.

We may conveniently define the editing part of this process to include all of the selection and cutting together of film either for dramatic or for technical reasons. We shall only touch upon this technique.

The re-recording part then includes the assembly of the sound onto film suitable for theatre use. Of course, this is only one of the purposes for which re-recording may be employed. Re-recording is to sound what process photography is to the picture, and most of the tricks available in process photography have their equivalent in sound re-recording.

Re-recording has four general purposes: To combine sound effects, to adjust loudness or quality, to secure additional or duplicate records, and to change the kind of record, as from disc to film or from push-pull track to single track.

These basic uses and their corollaries determine the nature of a re-recording channel. A stage recording channel uses one or more transmitters on the stage to initiate the electrical energy necessary for recording either on film or on disc. A re-recording channel merely replaces these transmitters with film or disc reproducers upon which records are played. The natures of these records may be very different. The dialogue may be from a film, music from film or several discs, and sound effects either from specially synchronized films or from loops of continuously repeating sound which can be appropriately mixed-in during a scene. Fundamentally, that is the only way in which the mechanics of re-recording differ from original recording. The methods of employing stage and re-recording channels are quite different, however, which becomes strikingly apparent when the entire subject of recording and reproducing motion picture sound is examined. Let us do that briefly.

1. THE GENERAL PROBLEM

We desire sound in the theatre which will seem so natural when heard with the picture, that the observers feel they are a part of the scene, or at least that they are viewing the production through a large opening in the end of the theatre. The specific technical requirements necessary to achieve this result are discussed elsewhere in this book.

2. EARLY SOLUTIONS

In the earliest sound pictures, a frequency band from about 100 to 5,500 c.p.s. was reproduced, the width of this frequency band being limited primarily by the theatre horns. Because of the limited frequency range and the use of simple horns, phase distortion was unimportant. The volume ranges of recording media were about 20 db from surface noise to maximum modulation, but, by accepting objectionable surface noise, 30 to 35 db of recorded volume could be secured in practice. There was little variety between scenes, whole reels commonly being made with one microphone placement. Music and sound effects

were recorded at the time the action was photographed and re-recording was usually not required or employed.

As time passed, scenes became shorter, several being re-recorded from intercut film or from a number of discs, to reel-length discs for release. Some pictures were released with the sound printed on the film from intercut original sound negative. Adding effects by re-recording was still uncommon. Level changes in the theatre, if made at all, were dependent upon "Cue Sheets," furnished to the theatre by the studio and according to which the projectionist made changes in the fader or amplifier gain.

Gradually the method of adjusting volume of variable density recording by changes in track density came into use. This is a valuable attribute of this system of recording which is still used. Planned use of this technique permits soft sequences, even in re-recorded track, to be printed darker than usual with the same resulting sound volume but with lower surface noise than normal density track having low modulation. In difficult re-recording, undesired level variations may occur in an otherwise acceptable re-recorded track. In the variable density system these can be corrected in the prints, but with variable area there is, at present, no easy cure. This feature makes variable density especially useful as a release medium.

3. INFLUENCE OF NOISE REDUCTION

Noise reduction systems became available for both variable area and variable density recording. These extended the useful volume range of variable density 6 to 10 db and of variable area about 12 db. At the same time the usefulness of re-recording was becoming more apparent. Re-recording permitted sequences to be shot in short scenes, or in different sets, or even on different days. Continuous background noises could be added later, and level and quality differences between original scenes corrected. Use of noise reduction on both original and re-recorded tracks reduced the surface noise to a somewhat commercial amount in spite of the increase due to re-recording. This increase is 3 db if the noise in the original record and in the re-recording medium are equal. Because of the random frequency spectrum of noise it combines in a root-mean-square manner. That is,

$$N_{recording} = [N^2_{original} + N^2_{rerecording}]^{1/2}$$

It is apparent that if either the original or the re-recording medium had even moderately increased noise reduction, the resulting total noise would be but little more than that of the noisier medium alone.

4. MODERN SOLUTIONS

The comparative freedom in production technique and the control of the release product, resulting from a policy of complete re-recording, caused several studios to adopt such a policy soon after noise reduction became available. The increasing use of process photography caused others to re-record more and more until now most productions are completely re-recorded. Conventional film recording methods were designed for printing the sound on the film beside the picture. They are still commonly used in this manner for theatre release. They had certain defects, however, which could be reduced by operation of two such tracks in "push-pull" (see Chapter III). If this push-pull track could be used for the original recording, the noise and distortion in the re-recorded film would be little more than that to which we had previously been accustomed in original records, and we could gain the great conveniences of re-recording at little quality cost. Not only has this been accomplished, but push-pull track has been adapted to the re-recording stage as well, and many modern theatres are equipped to reproduce such track. (See Chapter X.)

When this technique is augmented by the use of the squeeze mat, volume ranges of 55 to 60 db from maximum modulation in loud parts to minimum modulation (surface noise line) in soft parts are easily obtained in conventional theatres *without changing the theatre fader setting.*

If, in addition, the theatre can be equipped with the very inexpensive equalizer for reproducing track made with the complementary recording technique, 6 to 8 db more range is possible.

It seems improbable that more than 80 db range, according to the above definition, will ever be required.

5. MODERN DISTORTION REQUIREMENTS

It is most important to remember that sound, as currently recorded for picture use, passes through two acoustical, six mechanical, three electrical, six optical, and four chemical states; and twenty-four transformations between and within these states, at least twelve of which take place in the presence of superimposed mechanical movement.

Total quality distortions of 2 to 3% are detectable and little more is acceptable. Therefore, the total combined effect of all the partially non-linear elements such as vacuum tubes, iron-cored coils, film characteristics, and the transformations outlined above, must be within this amount of distortion. The greatest technical difficulty in re-recording is to keep errors and distortion so small that the combination of two of

them is not objectionable, when one at a time they may not even be noticeable.

6. RE-RECORDING IS A CREATIVE PROCESS

When you hear a famous violinist in Carnegie Hall you think of him only as a great creative artist—certainly not as a mechanical technician. Yet, if you were to spend hour after hour with him during practice, it is probable that you would become quite conscious of the meticulous placing of his fingers on the strings, his bowing, and even the kind of strings used on the violin. If, during all this time he *appeared* by necessity to be engrossed with technique, it is quite possible that when you went to the concert you would still think of him as a fine technician and of his violin as a mere mechanical tool. His concert would actually be just as beautiful a creation, but your point of view would have spoiled your appreciation of it.

It is fortunate that the audience seeing a finished picture has not seen it being rehearsed over and over again in the re-recording rooms. If the re-recorder is successful, the audience is not conscious of his technique but only of the result achieved. The director and producer watching the re-recorder work out the details, however, may think him and his tools very mechanical, for in spite of what is going on inside his head (the important part of re-recording), his hands are performing a multitude of mechanical operations, and his conversations with his assistants are in terms of machines.

7. RE-RECORDING TECHNIQUE

A re-recording monitoring room is a miniature theatre or projection room, with a picture screen, horn system behind it, and a large table on which are placed all the controls the re-recorder must manipulate.

The acoustic qualities of this room are important. It must sound as nearly as possible like a representatively large theatre. In fact, it is desirable for *all* projection rooms and theatres to sound alike. Then the product made or judged in one room will be equally good in the others as well as in the releasing theatres. These projection rooms need not necessarily be the same size, if each is properly made for its size.

The basic re-recording channel may be considered to be merely a projection channel exactly as in a theatre. A track played in the re-recording room should then theoretically sound the same as in the theatre. At a convenient place in this channel, as close to the horns as practicable, a recording machine is connected. If this recording operation does not distort, then the resulting re-recorded film (from which the process gets its name), should reproduce exactly what the re-recorder heard during the re-recording operation. It is most important that this

distinction be clear; that the re-recording operation records *what the re-recorder hears* and does not *necessarily* duplicate any original record. If the re-recorder is completely satisfied with the original, and does not modify it in any way, then the re-recording will duplicate it. If different recording media are used for the re-recording operation, and each has different degrees of inherent frequency distortion, individual recorder equalizers (see Chapter XVI) must be used which will permit each record to sound as nearly as possible like the monitoring. Inasmuch as the purpose is to record exactly what goes into the reproducing horns, it is desirable that there be as little circuit between the recording machine and the horns as is practicable, and that the amplifiers which supply the energy to the horns and to the recorder be stable in order that the relative levels and quality of the two may be constant. Photoelectric cell monitoring is unnecessary, for modern modulators are quite dependable and probably have less variation than a complicated monitoring device and circuit.

There are other methods of connecting and operating re-recording channels but none seems quite as simple as the one described above.

The re-recording mixer works primarily by ear, merely using his volume indicator to show the percentage of modulation of the film under different adjustments of his channel. The volume indicator is like a speedometer—you only look at it when you are about to exceed the speed limit. In ordinary driving you keep your eyes and your mind on the road, and there is no need for the speedometer.

The re-recording mixer has a number of volume controls, one for each record he is reproducing. He can insert equalizers into one or more of these circuits if the frequency characteristics are not as desired. With certain modern arrangements of volume and equalizer controls, one operator can usually mix five or six tracks and make necessary quality adjustments.

8. RE-RECORDING FOR DUPLICATE RECORDS

The operation of duplicating an original record, or transferring it to another recording medium with its sound quality and levels unchanged, has been described above. If more than one duplicate is needed, or if more than one type of new record is required, additional recording machines are connected to the circuit through their proper equalizers. If duplication into the same recording medium is desired it can often be done by photographic "duping," thus preserving any numbers and identification marks which may be on the original film, and assuring the same percentage modulation as the original (not always easily accomplished in re-recording).

The method of determining the desired recorder equalizer characteristic is simple. A recording is made of single frequency oscillator tones covering in steps the range from 50 to 8,000 cycles. Equal energy is supplied to the modulator amplifier at each frequency. This record then is re-recorded in a channel as described above, except that there is as yet no recorder equalizer. The original frequency record and the re-recorded record are then compared by measuring the voltage developed at each frequency when the records are reproduced into a voltmeter instead of into the usual horns. An equalizer is then designed which will cause the re-recording to match the original, which is inserted in the circuit and the test repeated as a check. The re-recording should now match the original. This equalizer should be adjusted precisely as departure of as little as 1 db in this range can be discerned by ear in music and dialogue.

The re-recording operation must also be free of phase and volume distortion. Phase distortion is limited by using as simple a channel as possible, with coils having high efficiency at frequencies even below the desired 50 cycle minimum, and equalizers which do not introduce unnecessarily large phase shifts at low frequencies.

Volume and frequency distortion are checked in a similar manner. A recording is made of a single oscillator tone, reducing its level in steps from full modulation to perhaps 30 db down. This record is re-recorded and compared with the original as described above. If either record fails to reproduce the original variations of oscillator level, volume distortion exists and must be corrected. The degree of distortion in the re-recording channel is indicated by the difference between the two records.

9. RE-RECORDING TO COMBINE SOUND EFFECTS

Re-recording is the sound equivalent of process photography, and ranges from adding a bit of crowd noise or a door knock in a single scene, to construction of a complete sound accompaniment to a picture sequence originally photographed silent.

The mechanics consist simply of placing desired sounds—dialogue, music, or other effects—at the proper place in two or more tracks, reproducing the tracks simultaneously, and controlling the loudness of each so that the desired effect is heard from the horns. The composite result is recorded as described above.

10. PREPARATION OF TRACKS

This editing operation is so closely related to re-recording that it should be considered briefly.

The sound recorded at the time the picture was photographed is cut simultaneously with the picture. Sound and picture should be

synchronized within two sprocket holes, and be kept in synchronism while editing by running them over sprocket wheels mounted on a common shaft. Other tracks, if required, are run over additional sprockets on the same shaft. If the sound recorded with the scene is not entirely adequate, suitable sound effects to make it so are selected from a film library and are cut into additional tracks at the appropriate places. It is desirable that a minimum number of tracks be used, but, on the other hand, it is inconvenient to jump from dialogue to sound effects to music in one track. It is customary to keep dialogue on one track, music on another, and sound effects on as many more as necessary.

As the tracks for each picture reel are assembled, their contents are outlined on a prepared form which serves to index the structure of the reel and to transmit or record editorial and technical instructions concerning its assembly. In one studio the reverse side of this form also serves as a record of all re-recording done on the reel. The sheets representing each reel during its various previews and release are always kept together, so that the entire history of the reel is available in one place.

The selection of effects from the library and their construction into the tracks requires much imagination, ingenuity, and care. The effects must be realistic and of the proper perspective, for if false they can destroy the whole illusion of a sequence. If illusion of reality is destroyed, the value of the motion picture is destroyed.

11. COMBINING THE TRACKS

The tracks can be mixed either for a realistic effect, or for a theatrical one. Realism puts quite strict limitations on the treatment. Theatrical effects usually utilize the exaggeration of certain sounds beyond realism, or the omission of realistic sounds. The combinations here are limitless.

Studio or producer policies may determine the general treatment. Some favor realism; others lean toward fantasy. In this connection, the manner of using music is important. If music is required by the action, it is a realistic effect—if merely for creating a mood, as an underscore during a dialogue sequence, it tends to destroy realism. The nature and treatment of the picture rather automatically determine how music should be used. If the story is being told by dialogue, sound effects must be subjugated. In other places,perhaps, the effects tell the story and may dominate the dialogue. Variety in treatment is refreshing if reasonable.

12. RE-RECORDING TO ADJUST LEVELS AND QUALITY

When sound is recorded in different sets, with different channels, under different monitoring conditions, many variations of level and

quality unavoidably result. In addition, actors may not speak quite the same in consecutive picture scenes, or they may even have a cold and distorted voice quality. One set may be heavily draped, and sound dull; another with mirrors all around will reflect sound about as effectively as light. Reverberant sound will seem loud though its energy content may be small.

The finally released picture must seem so smooth that it will appear to have been photographed and recorded in one long continuous operation. Re-recording is especially useful for this purpose. Level adjustment is inherent to it. The proper level for sound is determined very definitely by the action. The sound must appear to come from the picture on the screen. Because of the enlarged picture the proper loudness is somewhat greater than in life, usually 6 to 12 db. If too soft, it will seem remote, behind the screen, and if too loud it will be grotesque. The director's and actor's interpretation of the scene cannot be changed appreciably in re-recording by making the sound loud or soft. This would merely make the scene artificially unreal.

At present the setting of absolute loudness in the theatre must be determined by the theatre manager or projectionist. The re-recording mixer tries to help determine the proper loudness by the *range* of loudness he builds into the picture. If both whispers and shouts sound natural and convincing in the theatre, the sound is probably being well projected. ***Pictures are so constructed that if the dialogue is played at the proper level all effect and musical sequences will automatically be of the desired loudness without changing the theatre amplification.***

The standard of quality is always, "Does it sound like the real thing?" Most of the public have never heard a picture actor's actual voice, but everyone knows whether his picture voice sounds human and natural.

Most of us hear with two ears. Present recording systems hear as if with only one, and adjustments of quality must be made electrically or acoustically to simulate the effect of two ears. Cover one ear and note how voices become more masked by surrounding noises. This effect is overcome in recording, partially by the use of a somewhat directional microphone which discriminates against sound coming from the back and sides of the microphone, and partially by placing the microphone a little closer to the actor than we normally should have our ears.

This closeness tends to accentuate the low and high frequencies in the record, and make it both boomy and spitty.

Losses of high frequencies inherent to recording processes tend to help correct the spittiness, but the boominess remains, which in turn is accentuated by the loudness required for good illusion in the theatre. This boominess can be reduced in a number of ways, all of them amounting to "equalization."

Equalization can have many forms, as it is merely the use of reactive networks of one kind or another to give a system a desired relative transmission ability at different frequencies. The name originated from use in telephone circuits to counteract distortions which were otherwise unavoidable. This is still its primary use in recording. Since it is a counteractive distortion when so used, it is easy to see that it must be used intelligently or undesired new distortion will result. It can naturally be used to produce a controlled degree of distortion if desired.

Equalizing networks ordinarily are made with electrical reactances, but there are acoustical and mechanical reactances which can be used if convenient. Cavities in a microphone or loud-speaker and the mechanical resonance of light valve ribbons are examples. If a system is free from volume distortion, equalization can be introduced at any convenient point in the system with equal effectiveness. Amplifier systems, however, are not entirely free from volume distortion and this must be considered carefully in the design and use of equalizers. In addition, operating convenience often dictates the point in a system at which equalizers can best be used. A small fixed amount of low-frequency attenuation is sometimes used in the stage recording channel to reduce excessive boominess. In general, it is better practice to do corrective equalizing only while re-recording. Variations in quality can only be corrected after the picture is edited. In addition, the small group of re-recorders, working under standard, consistent monitoring conditions, closely in contact with release requirements, is better able to judge what treatment is required than a large group of stage mixers working under a variety of monitor conditions not particularly representative of theatres. Rather complex and bulky equalizer equipment is sometimes required, which can only be provided conveniently in the re-recording rooms. Variable equalizers may provide means to accentuate or reduce narrow or wide bands of frequencies throughout the entire frequency range. Fixed networks may be employed to counteract or produce specific, frequently encountered, distortions. Unusual problems may require the design and construction of special networks. In general, corrective networks should have the same nominal impedance as the circuit in which they will be inserted, and, as far as possible, should be of symmetrical constant resistance structure so that they can be used in tandem in any combination which may be required.

In the choice and use of corrective equalizers, a re-recorder can only be guided by ear. The nature and amount of correction must be determined by trial.

It is important to understand that phase and volume distortions cannot be corrected by attenuation equalizers. Reverberation is a phase effect—an echo, really, which cannot be removed from a record but which can sometimes be added. Volume distortion could only be corrected by introducing an inverse distortion in most cases, an almost impossible feat.

Another type of quality change sometimes utilized is to vary the pitch of a sound, which is accomplished by operating either the reproducer or the recorder at other than normal speed. This naturally affects the duration of the sound in proportion to the speed and pitch change. However, the harmonic relationship of overtones are not upset, so moderate changes do not cause unintelligibility in dialogue or harshness in music. Such a change does modify the quality of the sound in addition to changing its pitch. A masculine voice raised an octave by this means would sound feminine.

13. SUMMARY

We are painting a picture. All of the mechanics we employ, whether acting, directing, photography or recording, are the brushes with which the thought colors are spread on the theatre screen. If one can get a better effect by smearing the paint on with his thumb than by using a more conventional tool, there is reason for doing it. Progress results from new tools developed to meet new needs.

Chapter VI

MICROPHONES

By L. E. CLARK

1. TYPES

A microphone is an electro-acoustic transducer, actuated by either the pressure produced by, or the velocity of, sound waves, which converts acoustic energy into electrical energy.

Theoretically, the current wave produced in the electric circuit should follow exactly the pattern of the pressure wave, and any differences in the two waves are the result of some deficiency in the microphone. The microphones most commonly used in studio work are, in historical order: carbon, condenser, dynamic or moving coil, ribbon or velocity, and crystal.

These can be divided into two groups: pressure operated devices, including the carbon, condenser, dynamic, and crystal; and the velocity operated device, of which the ribbon is an example. Of the first group, the carbon, condenser, and crystal are tuned; the dynamic, untuned.

The condenser, dynamic and ribbon are the most commonly used.

The output of these microphones varies from about minus 40 to minus 80 db, compared to a reference level of one volt open circuit per bar of acoustical pressure (1 bar equals 1 dyne per sq. cm.). Such a small output requires a great amount of amplification, and in some cases the first, or microphone amplifier, is actually contained within the microphone housing itself to reduce losses due to amplifier leads.

The output impedances vary from a few ohms to several thousand ohms. The high impedance instruments must have their amplifiers in the same housing to reduce losses, but the output from the low impedance microphones can be carried over a considerable distance.

2. CARBON MICROPHONE

The carbon microphone is the oldest type, operating upon the principle that the resistance of a mass of carbon granules varies with the

pressure applied. A direct current passes through the carbon, and variations in pressure cause direct-current variations which theoretically follow the pressure changes. A transformer is inserted in the microphone circuit which allows only the alternating portion of the current to be delivered to the amplifier circuit.

Commercial units of this type usually employ two such masses of carbon (the so-called double-button carbon microphone) to reduce amplitude distortion.

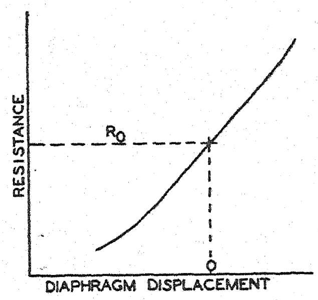


Figure 42 — Diaphragm displacement of a carbon microphone.

Figure 42 shows a graph of the resistance of the circuit plotted against the position of the diaphragm. At zero displacement the resistance has some value R_0 and as the diaphragm is displaced, distortion results as the curve is not linear. The double-button microphone attempts to correct this by superimposing an opposite curve of the same type upon the first in a push-pull circuit, which reduces not only the distortion but also the sensitivity.

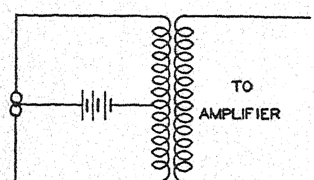


Figure 43-A — Schematic of carbon microphone circuit.

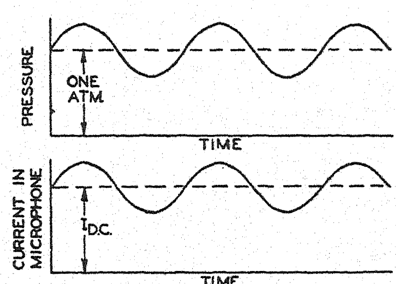


Figure 43-B — Electrical wave in microphone circuit showing the corresponding pressure wave.

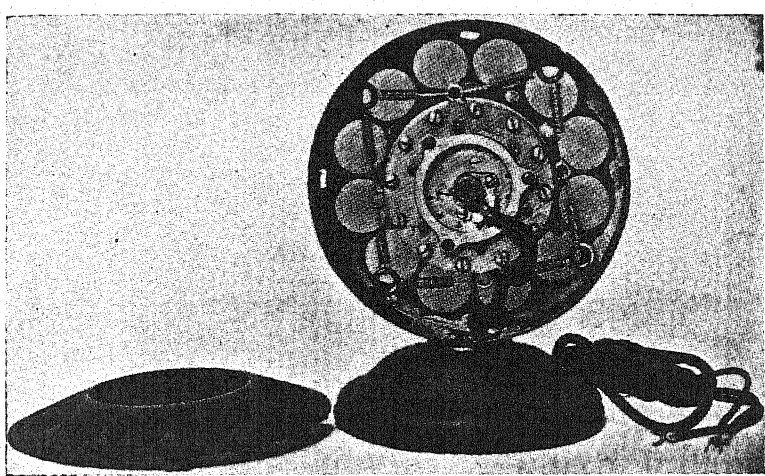
Figure 43-A shows a schematic circuit of such a microphone. Figure 43-B gives a simplified pressure wave and the corresponding electrical wave if no distortion is present. The current I delivered to the amplifier is

$$I = I_M - I_{DC} \qquad (17)$$

where I_M is the total current at any instant through the microphone circuit, and I_{DC} is the no-signal or unmodulated direct current.

Figure 44 shows this microphone as commercially manufactured.

The disadvantages of this microphone are: High noise level, general instability of frequency characteristic, and tendency of the carbon granules to pack together.



—*Courtesy Electrical Research Products, Inc.*

Figure 44 — Carbon microphone.

The hiss or background noise is due to the great number of contacts presented by the large number of granules of carbon.

The sensitivity depends upon the condition of the carbon, and is affected by vibration, position, handling, etc., and so the microphone is very apt to lose its calibration.

This microphone is not used in the studios because of stability and noise-level requirements which it cannot meet.

3. CONDENSER MICROPHONE

The condenser microphone derives its name from the fact that its operation depends upon the variation in the capacitance between two plates. One plate, the diaphragm, is movable and the other is fixed, the two forming a variable condenser in which the distance between the plates is the variable element. A direct-current potential is applied to the two plates through a high resistance, and as the diaphragm vibrates and produces a corresponding change in capacity, voltage variations appear across the series resistor. This voltage can then be amplified, and since the resistor across

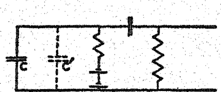
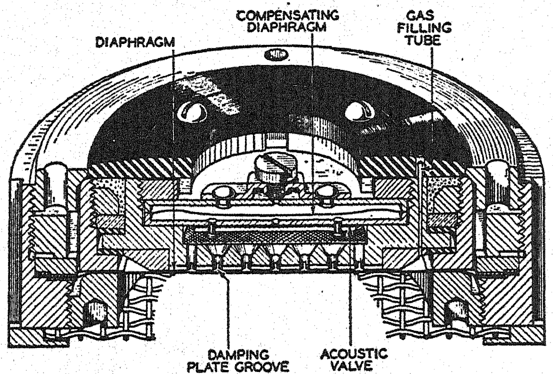


Figure 45 — Schematic diagram of a condenser microphone.

which it is developed is very high, the first amplifier tube is usually coupled through a small capacitor to the resistor.

In microphones of this type now in use the impedance is high, and the amplifier immediately associated with the microphone is in the same housing.

Figure 45 shows a schematic wiring diagram of a condenser microphone, using a resistance coupling, while Figure 46 shows a cross-section of a typical condenser microphone.



—*From "Radio Engineering," by Frederick E. Terman (McGraw-Hill Book Co., Inc.).*

Figure 46 — Cross-section of condenser microphone.

The diaphragm is duraluminum and stretched so that its resonant point is well up in the recording band. The overall diameter is only about three inches, and as a consequence the plates must be relatively close together to have an appreciable capacitance. The gas trapped in the small space between the plates acquires a stiffness and viscous damping and its action results in raising the frequency of resonance. The effect of this resonance is minimized by designing the microphone so that the resonance mentioned above occurs at the extreme upper end of the frequency range.

In order that the stiffness of the trapped air be relatively constant with frequency, the usual design divides the back plate into a number of small sections by means of crosscut grooves or concentric rings.

Equalization is provided to take care of variations in atmospheric pressure. This is shown in Figure 46 as the compensating diaphragm.

(a) Cavity Resonance

The condenser microphone is so constructed that there is a cavity or air pocket in front of the diaphragm (see Figure 46). At the fundamental frequency of vibration of this cavity, the pressure may build up to twice the value it would have if this cavity were not present.

The resonant point is usually in the neighborhood of 3,500 c.p.s., depending on the dimensions of the cavity, and has a magnitude of about 5 db at this frequency for waves approaching near the axis of the microphone.

The magnitude of the cavity resonance has been reduced in some microphones by modifying the housing so that the diaphragm is flush with the face of the microphone.

(b) Pressure-Doubling Effect

If the frequency of a sound is high and its wave length small compared to the size of the diaphragm, part of the sound wave is actually stopped, and the pressure at the diaphragm increased. In the high-frequency range, the sensitivity of the microphone may be increased as much as 6 db. The actual frequency at which this increase takes place depends upon the size of diaphragm and mounting.

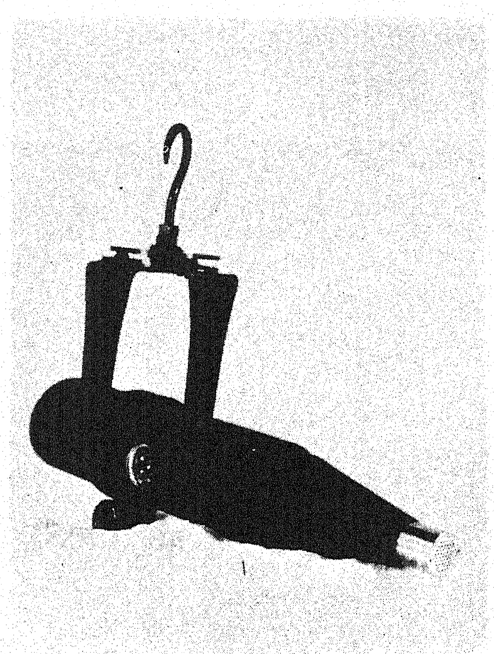


Figure 47 — Miniature condenser microphone (ERPI—D-99848 transmitter with amplier).

The effects of cavity resonance and pressure doubling on the response characteristic of a condenser microphone are shown in Figure 58. These effects are additive and give rise to the peak in the region of 3,000 cycles;

pressure-doubling contributing approximately 6 db, and cavity resonance 5 db.

The condenser microphone has a higher impedance, a lower sensitivity, and a much lower noise-level than the carbon type, and is affected by cavity, diaphragm, and air-pocket resonances (the last two being damped), and, in sizes now commercially available, by pressure-doubling. Its response characteristic is comparatively stable, and it is not affected appreciably by temperature, humidity or ordinary handling.

A smaller improved type of condenser microphone is now available commercially. The diaphragm, built flush with the face of the housing, has a diameter of approximately 0.6 of an inch.

Figure 47 shows the new type miniature condenser microphone (usually termed Baby Condenser) with its associated amplifier. The size of the microphone element itself is about eight-tenths of an inch in diameter by approximately an inch in length.

The reduction in size shifts the cavity resonant peak from the 3,000 cycle neighborhood of larger condenser microphones to the high-frequency range at approximately 10,000 cycles.

The efficiency is approximately the same, compared to the larger microphone, as the proportion of "dead" to "active" capacitance is much lower, and for this reason the voltage on the grid of the first amplifier tube is the same in both cases and no increase in noise results.

Still further reductions in size have been considered in an attempt to improve reflection and phase difference effects but the results were of negligible importance within the audible frequency range.

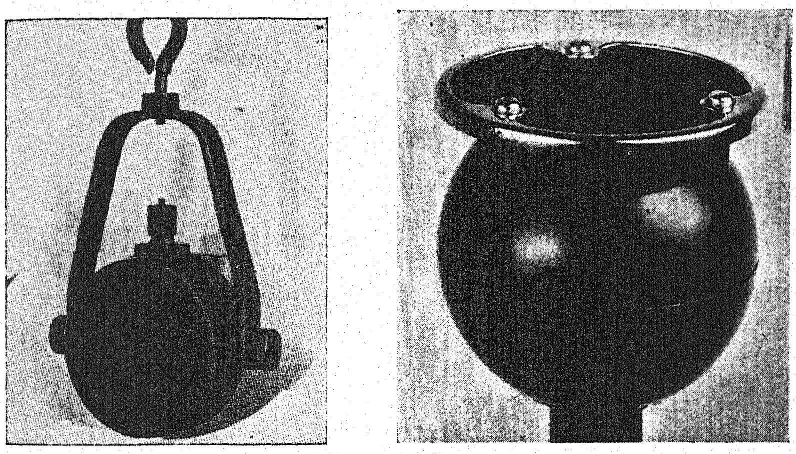
A flatter response curve is obtained from this smaller microphone (see Figure 58) due to the decrease in diameter and the elimination of the cavity.

4. DYNAMIC OR MOVING COIL MICROPHONE

This microphone operates upon the principle that a coil, when moved through a magnetic field, will have an e.m.f. induced in it. It consists mainly of a diaphragm to which a coil is rigidly fixed, the diaphragm vibrating in response to the sound waves striking its surface and causing the coil to vibrate in like manner, cutting magnetic lines of force.

By the very nature of this design, the mass of the moving parts is relatively large, and acoustic networks must be used in order to give a flatter response.

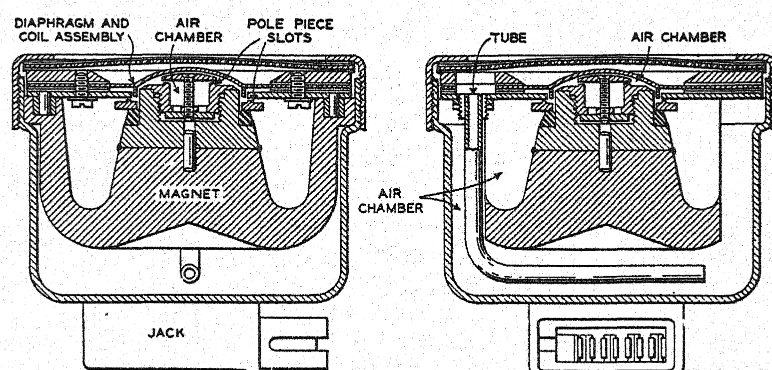
Figure 49 shows a cross-section of a dynamic microphone. It will be noticed that the diaphragm is crowned and the voice coil placed at the edge of the crown, and that there is an acoustic network coupled to the diaphragm.



—*Courtesy Electrical Research Products, Inc.*

Figure 48 — Dynamic microphones.

This microphone has a higher sensitivity than the ribbon, a low-hiss characteristic, low impedance, and resonant peaks which are



—*Courtesy Electrical Research Products, Inc.*

Figure 49 — Cross-sectional view of a dynamic microphone.

acoustically damped. The response characteristic (see Figure 58) is good, but not always uniform from one microphone to another of the same type.

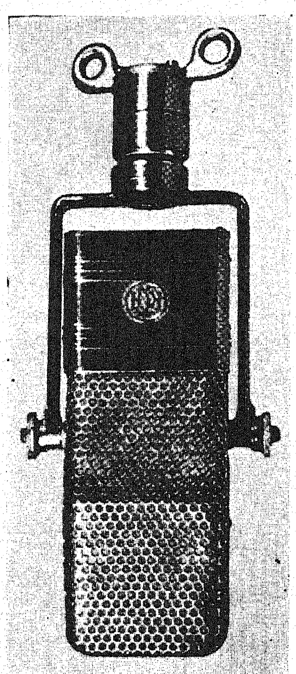
5. RIBBON MICROPHONE

The ribbon microphone consists of a very light, corrugated metallic ribbon suspended under negligible tension in a magnetic field and freely accessible to air vibrations from both sides. The ribbon vibrates in the magnetic field under the action of the difference in pressure existing between the two sides and this vibration induces an e.m.f. in the ribbon. Although the generated e.m.f. of this type microphone is low, its impedance is low and, therefore, the efficiency is of the same order of magnitude as the condenser and dynamic microphones.

In order that the ribbon or diaphragm move, it must be actuated by the pressure of the sound wave. This is true for all types of microphones, including the ribbon, or velocity microphone, which is operated by the pressure difference between the front and back. Strictly speaking, it should not be called a velocity microphone, but a pressure-gradient microphone.

Although the sensitivity of this type microphone is low, its amplifier need not necessarily be in the same housing, as its impedance is also very low.

Its response characteristic is the best of all type microphones with the possible exception of the crystal, but there is a characteristic middle-frequency hum which is troublesome when the microphone is used for dialogue recording, but when used for music (its most frequent use in the studios) this hiss is not apparent.



—Courtesy RCA Manufacturing Co.

Figure 50 — Ribbon microphone.

It is, however, easily overloaded, and sudden sounds such as gun shots, explosions, etc., may blow the ribbon entirely out of the air gap.

Figure 50 shows a ribbon microphone.

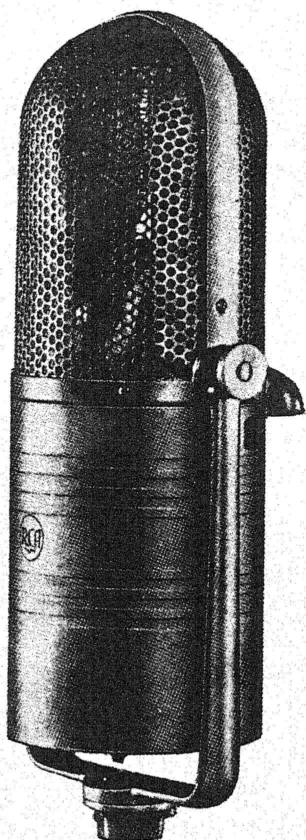
6. UNI-DIRECTIONAL MICROPHONE

The uni-directional microphone is a microphone that is partly pressure and partly pressure-gradient operated. The moving element

is again a ribbon, but it is divided into two sections. One section acts as a pressure-gradient microphone as described above. The other section is closed at the back by means of an acoustic labyrinth which removes the pressure from the rear. Thus the lower section of the ribbon is a pressure-operated device.

The output voltages of the two sections of the ribbon are in series. For sounds coming from the front, the developed voltages add, but for sounds coming from the rear, these voltages are 180° out-of-phase and, therefore, cancel. Figure 57 shows the directional characteristic of (a), a pressure-gradient microphone; (b), a pressure microphone; and (c), the combination of these to make a uni-directional microphone.

The uni-directional microphone is used where it is desired to discriminate against sounds coming from the rear of the microphone and which would be picked up by a pressure-gradient microphone. It also has a wider pick-up angle.



—*Courtesy RCA Manufacturing Co.*

Figure 51 — Uni-directional microphone.

7. CRYSTAL MICROPHONE

The crystal microphone is one of the latest types to be developed (Figure 53).

Its operation depends upon the principle that certain crystals have piezo-electric properties, that is, the crystal develops electrical charges on certain surfaces when subjected to mechanical stress.

The usual construction is such that the sound pressure causes the crystal to be bent, and as a consequence of the strain, voltages are produced which are proportional to the pressures of the sound waves.

The crystal microphone has a very good characteristic, provided there are no temperature changes during recording. The resonant peaks are above the audible range, and the inherent hiss is low. However, the

impedance is very high, sensitivity low, and the microphone is unstable when subjected to humidity and temperature changes.

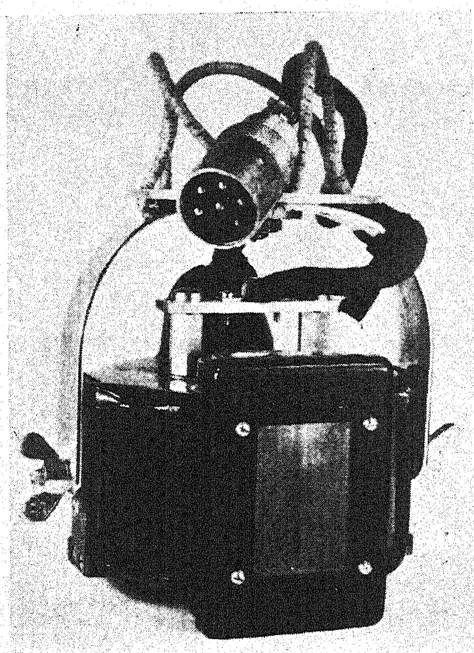
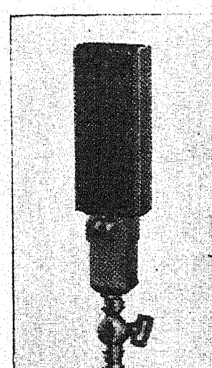


Figure 52 — Siemen's uni-directional microphone.

For these reasons, this type microphone is excellent as a test microphone, but is only used in production on musical stages where temperature and humidity conditions can be rigidly controlled.

The sensitivity may be increased by the use of a diaphragm mechanically linked to the crystal. Through this lever action obtained, a greater force, for any given sound pressure, is applied to the crystal surface. This type is not used in the studios as this construction results in resonant peaks in the audible range.



—Courtesy Brush Development Co.

Figure 53 — Crystal microphone.

The output of these microphones can be carried through a concentric cable to a separate amplifier (maximum of 50 ft.) resulting in a reduction in output level without varying the frequency characteristic.

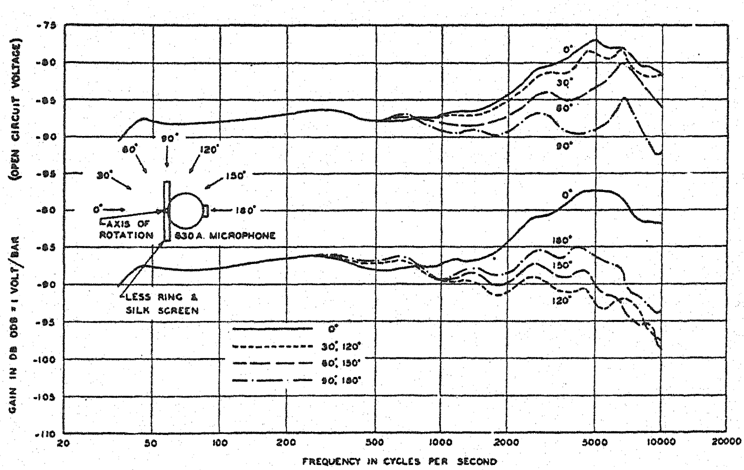


Figure 54-A — Directional response of a small dynamic microphone.

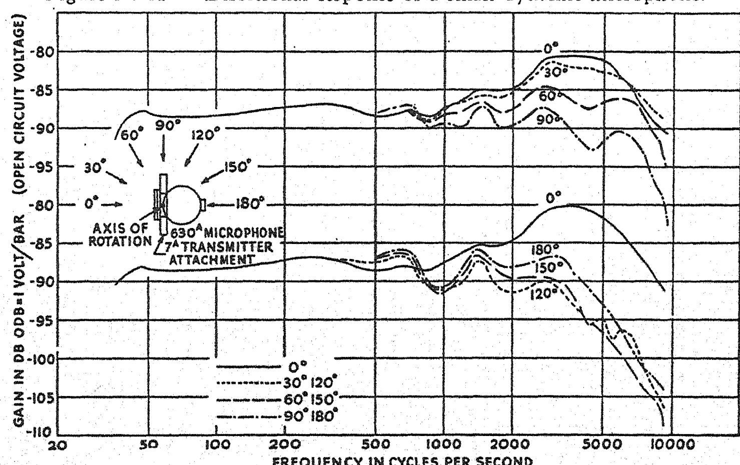


Figure 54-B — Directional response of a small dynamic microphone with baffle attachments.

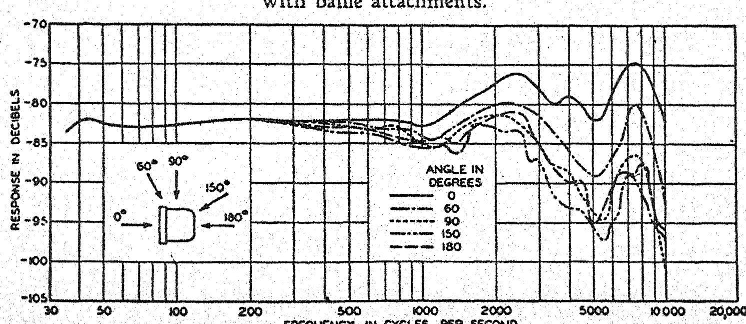


Figure 54-C — Directional response of a larger dynamic microphone.

—*Courtesy Electrical Research Products, Inc.*

8. DIRECTIONAL EFFECTS OF MICROPHONES

The directional effects of pressure-operated microphones depend upon the size of the microphone, because as the size of the diaphragm decreases, the microphone tends to act more as a "point" receiver of sound, and the effect of pressure doubling decreases rapidly.

Figure 54 shows the typical response at various angles of incidence of dynamic microphones of different sizes. Figure 54-A gives the response of a microphone, the diameter of which is approximately 2½", while Figure 54-B gives the response for this microphone when equipped with attachments to give a directional quality. Figure 54-C gives the response of a microphone with a diameter of 3¼".

The directional effects of the pressure-gradient devices depend upon the angle of incidence of the sound wave with respect to the ribbon surface. Figure 55-A shows the response of such a microphone at

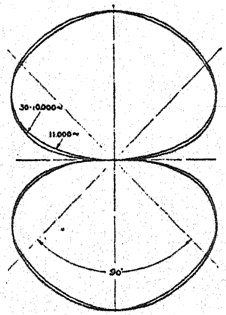


Figure 55-A.

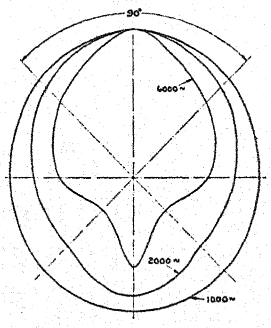


Figure 55-B.

—*Courtesy RCA Manufacturing Co.*

various angles of incidence. If the sound wave is approaching the ribbon normal to its surface, the air particles have a much greater surface to act upon, and consequently affect the ribbon to a much greater extent than if approaching from an angle on either side of the normal.

From Figure 56 it may be seen that when a sound wave is approaching from the direction indicated, the effective surface of the ribbon is no longer S but S', and the result is a weaker response from the ribbon. Figure 55-A shows that this effect is much more critical as the sound approaches nearly parallel to the plane of the ribbon.

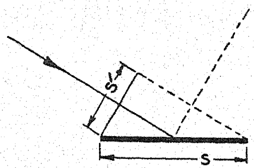
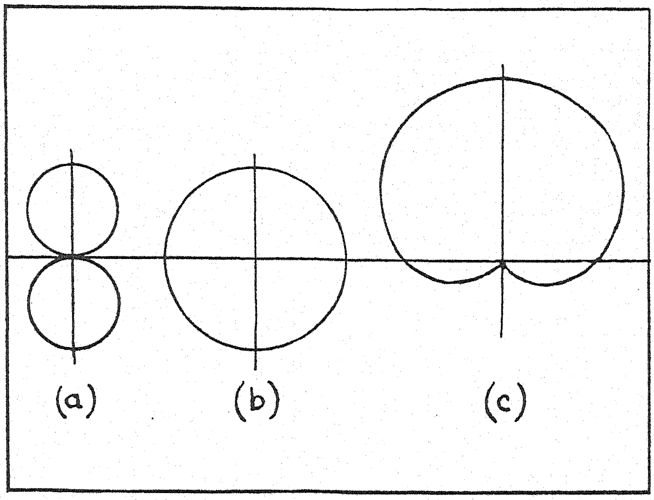


Figure 56.

Figure 55-B shows typical directional characteristics for pressure-operated microphones.

Figure 57 shows the directional effect of the uni-directional microphone. The velocity effect (a) combines with the pressure effect (b) in an additive manner on one side of the microphone axis, and in a neutralizing manner on the other side of the axis, giving the combined result as shown at (c).



—*Courtesy RCA Manufacturing Co.*

Figure 57 — Comparison of directional characteristics of velocity (a), pressure (b), and (c), a combination of pressure- and velocity-operated devices.

The requirements for microphones for motion picture production are such that they seldom can be placed in the ideal position for sound pickup, as camera lines, microphone shadows, etc., must be taken into consideration when the microphone placement is being made. As a consequence, the microphone position is in most cases a compromise. This results in the use of microphones which are not too critical in their placement and which are fool-proof and uniform under operating conditions, even though under laboratory conditions they might not prove to be technically the best instruments.

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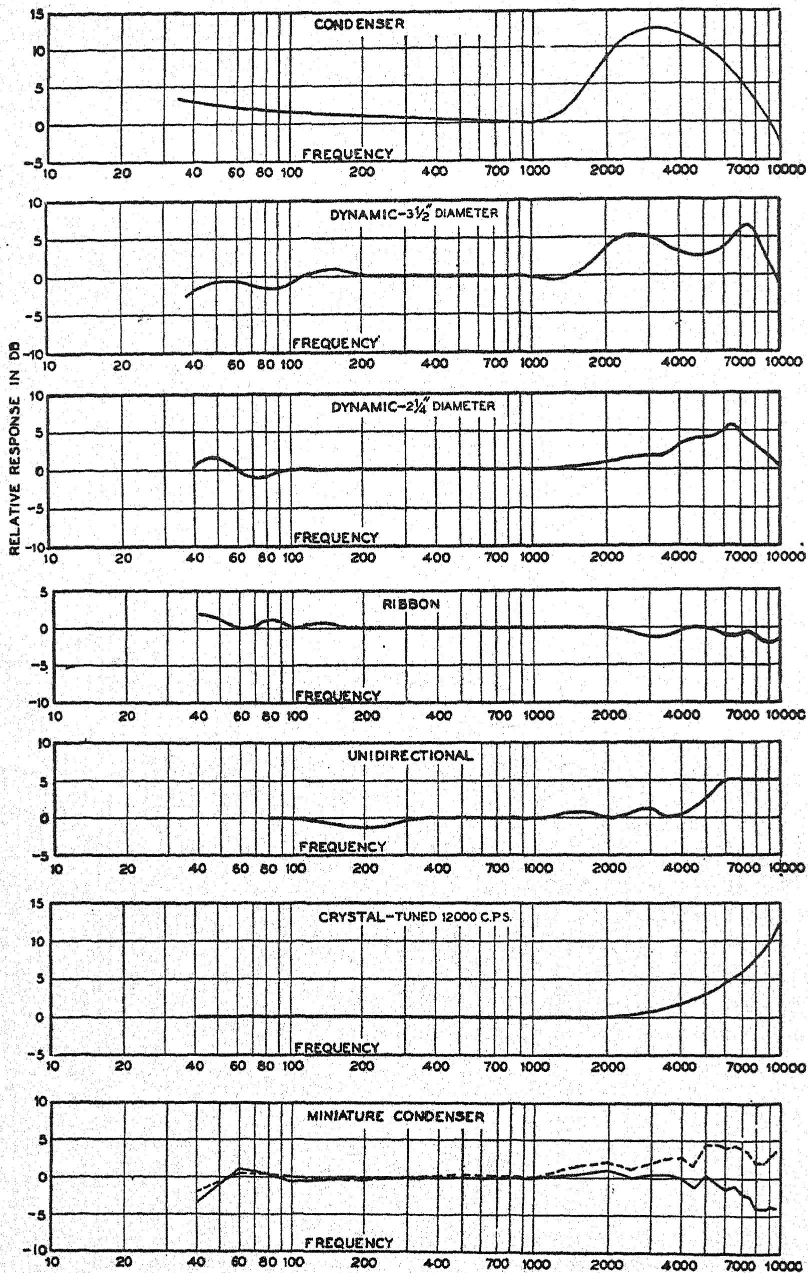


NOTE—All measurements made at 0° (normal) incidence (see Figure 54) except for Miniature Condenser. For Miniature Condenser dotted line curve at 0°, full line curve 90° incidence.

Figure 58 — Typical characteristics of commercially manufactured microphones.

Chapter VII

HEADPHONES AND LOUD-SPEAKERS

By L. E. CLARK *and* JOHN K. HILLIARD

Headphones and loud-speakers are electro-acoustic transducers which receive power from an electrical system, convert it to mechanical power, and then deliver it to an acoustic system. This transfer is usually made by the electrical current reacting with a magnetic field and causing an armature or coil of wire to vibrate in step with the current. The armature or coil is mechanically connected to a diaphragm which in turn vibrates the air and produces sound.

1. HEADPHONES

Headphones, as their name implies, are used against the ear, and produce variations in pressure in the small volume of air trapped in the ear between the ear drum and the diaphragm of the phone.

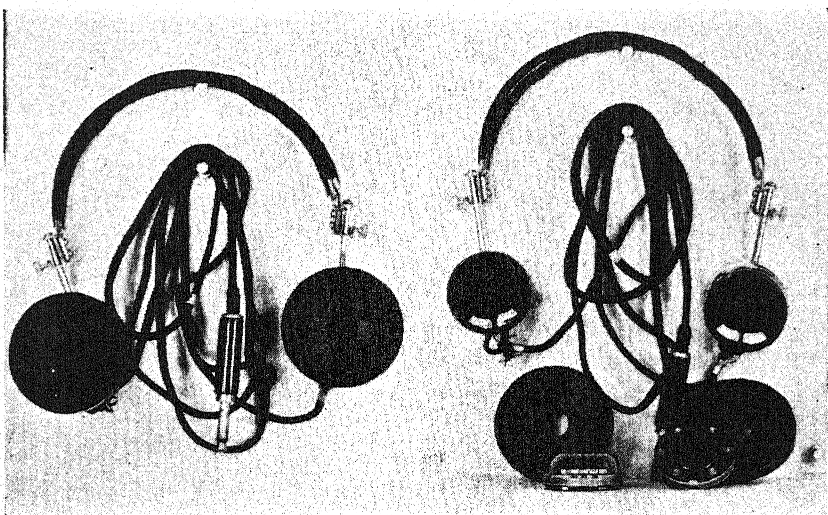


Figure 59 — Typical headset used in the studios, left view assembled, right view unassembled (ERPI No. 705).

The electro-magnetic telephone receiver, which was the earliest magnetic-type headphone, consists of an iron diaphragm vibrated by the variations in a magnetic field produced by the electric current in coils

placed over the pole tips of a permanent magnet. This receiver has a very poor frequency response characteristic, reproducing only the middle tones because of the sharp resonance of the diaphragm.

A so-called high-fidelity headphone has recently been designed which employs a moving coil, or electro-dynamic, telephone receiver, similar in construction to the moving coil microphone. The microphone and receiver cannot be used interchangeably, however, as their design requirements are different.

In the electro-dynamic receiver, the diaphragm is actuated by the force developed between a magnetic field and the electrical current flowing through the voice coil which is rigidly coupled to the diaphragm. As in the case of the microphone, the frequency response is smoothed out by an acoustic network coupled to the diaphragm.

Another type of high-fidelity headphone employs a straight conductor, or ribbon, which serves the dual purpose of carrying the actuating current and performing as a diaphragm. Its response is smoothed out by an acoustic network coupled to the ribbon.

High-fidelity headphones must be designed with a consideration of the effect of the acoustical properties of the ear, such as the air chamber in front of the ear drum, etc., and of the acoustic response of the ear, which has been discussed in Chapter II. Account must also be taken of a nominal amount of leakage which occurs between the phone unit and the ear.

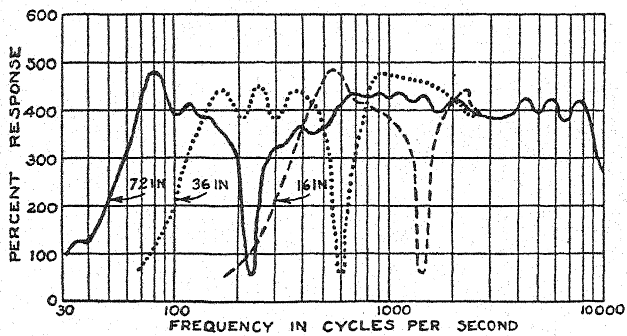
2. LOUD-SPEAKERS

Loud-speakers are electro-acoustic transducers designed to handle relatively large amounts of power, and to distribute the reproduced sound over a large area. They differ from headphones in that they must be able to produce more acoustic power and must radiate it into a different type of receiver. Fortunately, they are not subject to the same space and weight limitations which must be imposed upon headphones. In general, loud-speakers are unable to radiate low frequencies efficiently because loss of coupling occurs between the diaphragm and transmitting medium. For this reason they are usually used in conjunction with a baffle or horn.

The earliest speakers were electro-magnetically operated and were very similar to the electro-magnetic receiver. Some used a large cone for a radiating surface while others were merely powerful telephone receivers coupled to horns. They were subject to the same deficiencies in response as were the electro-magnetic receivers.

Most present day loud-speakers are of the electro-dynamic type and employ either a diaphragm or a cone. For radio receivers and other low

power applications, the speaker is usually used in conjunction with a flat baffle to increase the low-frequency radiation. As the cone vibrates, the waves sent out from the front and from the back are 180° out-of-phase, and there is a tendency for the pressure built up in front of the cone to



—*From "Applied Acoustics," by H. F. Olson and Frank Massa (P. Blakiston's Son & Co., Inc.).*

Figure 60 — Percentage response-frequency characteristic of 8 inch cone loud-speaker mounted in various sizes of square baffles. Measurements in free space, microphone distance equal to 10 feet.

be relieved by a flow of air into the corresponding region behind the cone instead of being radiated outward. This effect is of more importance for low frequencies where the path between the front and back of the cone is an appreciable part of the wave length. The flat baffle increases this path, prevents the air flow, and consequently increases the low-frequency radiation from the cone. Figure 60 shows the frequency response of a cone speaker in baffles of different sizes.

For motion picture theatre applications and other applications where large amounts of high quality acoustic power are required, horns or directional baffles are employed because of their much better performance and their ability to direct the power into useful areas. The horn functions much in the same manner as the flat baffle in preventing pressure equalization and at the same time results in higher efficiency and desirable directional characteristics. It can be thought of as an acoustic matching transformer that couples the relatively heavy diaphragm to the lighter air.

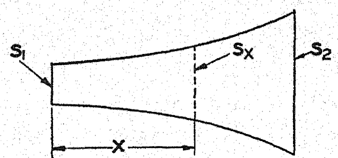


Figure 61 — Exponential horn.

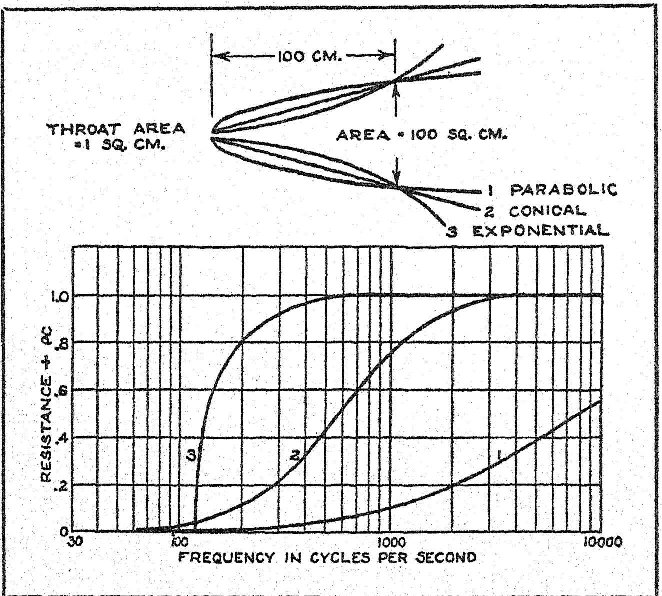
It is general practice to use horns with an exponentially increasing cross-sectional area, i.e., the area S_X at a distance, X, from the throat

is (see Figure 61),

$$S_X = S_1 \epsilon^{MX} \qquad (18)$$

where S_1 = Area at throat
S_X = Area at distance X from throat
X = Distance from throat
ϵ = 2.718
M = Constant which determines the taper of the horn.

The theory of horns shows that, while a horn of almost any shape will cause the impedance match to be attained at a low frequency, a horn of exponential shape will cause it to be attained at the



—*From "Applied Acoustics," by H. F. Olson and Frank Massa (P. Blakiston's Son & Co., Inc.).*

Figure 62 — Throat resistance of infinite parabolic, conical and exponential horns, having the same throat opening and the same cross-sectional area at 100 centimeters from the throat.

lowest frequency possible in any design. The impedance increases to a maximum value much more rapidly in an exponential horn. Figure 62 shows the throat resistance of different types of horns.

The resistance at the throat is given by

$$R = \frac{\rho C}{S_1}\sqrt{1 - \left(\frac{MC}{2\,\omega}\right)^2} \qquad (19)$$

where ρ = density of air
C = velocity of sound
$\omega = 2\,\pi f$

For frequencies below $\omega = MC/2$, the resistance is zero, and the horn acting as a high-pass filter can radiate no power. Therefore, in the design of a practical horn, it is necessary to choose M so that the cut-off frequency of the horn is well below the lowest frequency to be radiated.

In addition, consideration should be given to the size of the mouth; a usual criterion being that its perimeter should be comparable to the wave length of the lowest frequency it is to pass. The size of the throat is governed on one hand by the largest practical size of cone or diaphragm and on the other by the restrictions imposed on the length of the horn by space requirements in the theatre, etc.

The remainder of this chapter is devoted to a description of the speaker mechanisms and horns that are used in present day high quality reproducing systems.

TWO-WAY HORN SYSTEM*

3. INTRODUCTION

The present investigation was undertaken with a two-fold purpose in mind; first, to study thoroughly the more important types of extended range loud-speaker systems in current use, and second, to develop if possible, a system which would combine practicability for theatre use with as great an improvement in quality and efficiency as could be obtained without greatly increased cost. The first objective necessarily involved an effort to learn as much as possible of the "why" as well as the "how" of the systems and individual speakers studied, while the second led to considerable investigation of certain aspects of loud-speaker design, some of which—at least in the literature of the subject—seem not to have been sufficiently emphasized in the past.

Any investigation of as wide a scope as the present one inevitably furnishes many facts not pertinent to the main issue, but useful in other fields. The main body of the paper has been written with the problem of the reproduction of sound for motion pictures ever in mind, and should be read from that viewpoint. It is felt, however, that the results referred to may form a definite contribution to other fields, such as public address work and home radio.

4. SOUND REPRODUCING SYSTEMS FOR MOTION PICTURE THEATRES

The art of modern reproduction of sound in motion picture theatres is now about eight years old. During this time there has, of course, been

* Reprint from the Technical Bulletin of the Research Council of the Academy of Motion Picture Arts and Sciences, March 3, 1936.

considerable improvement, but there has been only one major change in the standard theatre installation. This change was the adoption of the "Wide Range" and "High Fidelity" systems after 1933. The principal modifications involved were: First, a partial fulfillment of greatly needed increase in amplifier carrying capacity; second, the adoption of speaker systems which provided for the division of power between two or more groups of speakers, each operating over a limited-frequency range; and third, improvements in the sound head which reduced flutter. While these improvements considerably raised the standard of reproduction in the theatre, it was felt that the loud-speaker system still constituted the principal limitation to naturalness of reproduction. An investigation was accordingly made to determine whether a speaker system could be developed which would economically replace the present systems while providing the much needed increase in fidelity. This was found to be the case, and it is the purpose of the present paper to describe this system and the results obtained with it, and to compare it with previous systems.

Since it was not known how great a departure from a full range linear response could be tolerated for the purpose in hand, it was considered advisable to start with a system as near this as so far achieved, even though the form of apparatus available by its size and cost would prohibit its use for theatre installations. From this it was determinable how much deviation was allowable and necessary in order to obtain a commercially practical system. Such a linear system was made available, and a series of tests led to the following specifications which were found to be adequate for theatre reproduction, taking into consideration further developments in recording which may be expected within the next few years.

5. SPECIFICATIONS

Flat Overall Frequency Characteristic. The system shall not deviate by more than plus or minus 2 db, from 50 to 8000 cycles, over the entire angle of distribution within ten feet of the mouth of the horn.

High Electro-Acoustical Efficiency. It shall approach fifty per cent in order that the required amplifier capacity will not be too great.

Volume Range. The volume range shall be at least 50 db and preferably 60 db.

Reasonable Cost.

Absence of Transient Distortion and "Fuzziness." The electro-acoustical transducer shall be of such construction that it will not generate objectionable harmonics up to the peak power required, and

the phase delay between units shall be such that the sound will be equivalent to that coming from a single source.

Suitable Angular Distribution Characteristics. The sound shall be radiated through a horizontal angle as great as 110 degrees and a vertical angle of 60 degrees, with nearly uniform response at all positions.

Reasonable Compactness and Portability. Low weight.

Amplifier Capacity. The installed amplifier capacity shall be such that one acoustic watt per one thousand square feet of floor area can be delivered when the auditorium is adjusted for optimum reverberation time.

A system which will conform to, or exceed, these specifications has now been developed, and can be constructed at moderate expense.

In order to take advantage of these characteristics it has been found that when film is reproduced over a system such as this, it is necessary to keep the flutter from the sound head no greater than 0.1 of 1 per cent. Although the problem of flutter has been satisfactorily solved and heads are commercially available which will pass the 0.1 of 1 per cent flutter specifications, it should be pointed out that by far the largest majority of heads in use today will not meet this specification.

6. POWER AND FREQUENCY REQUIREMENTS

The history of the electrical reproduction of sound has been one of continual increase in amplifier carrying capacity, and in this respect, the theatre installation is no exception. Originally, output powers from 2.5 to 12 watts were considered adequate for most houses. With the advent of the later systems now in use, these powers were recommended to be increased from 3 to 6 db, depending upon the size of the house. It has been found from this investigation that it is both practical and eminently desirable to make a further increase of at least the same amount. The figure given of one acoustic watt per one thousand square feet of floor area is felt to be the minimum which will do justice to the advanced conception of reproduction of records produced with modern recording technique. It is of interest to note that this figure can be achieved allowing for considerable latitude above this point without danger of mechanical damage to the units.

The advisability of extending the frequency range of a reproducing system must be determined by balancing the gain in naturalness obtained by the extension, against the resulting increase in noise and extraneous sounds. In the present state of the recording art, a characteristic flat to 6,000 cycles is the least that will do justice to the film; an extension to 7,000 or even 8,000 cycles is advisable, and a furhter

extension is not. This is so because a further extension becomes of less and less value, due to the decreasing sensitivity of the ear and the small amount of energy in this region, and especially because above 8,000 cycles, noise, flutter and harmonics due to recording deficiencies become decidedly the limiting factor. Incidentally, since practically all recording systems include a low-pass filter with a cut-off in the neighborhood of 8,000 cycles, there is nothing on the film at high frequencies to be reproduced.

Once the high-frequency limit is chosen, the low-frequency limit is automatically fixed. It has been found that for ideal balance the product of the two cut-off frequencies must be fairly close to 400,000, so that for an 8,000 cycle upper cut-off, the lower becomes 50 cycles.

7. HIGH-FREQUENCY HORN

One of the principal limitations of present theatre installations is bad directional characteristics. The plain exponential horn has a directivity which varies with frequency; low-frequency sound is projected fairly uniformly over a wide angle, but as the frequency is increased this angle decreases rapidly until at frequencies of several thousand cycles practically all of the energy is emitted in a narrow beam. The result of this is that the reproduction becomes very "drummy" or "bassy" for that portion of the audience whose seats lie well off the axis, while the opposite is true for seats located directly on the axis. In the present system this effect is eliminated by using a radiating system for the high-frequency unit which is composed of a cluster of small exponential horns, each having a mouth opening of approximately sixty square inches. These individual units are stacked in layers to form a large horn, the mouth opening of which is spherical in shape. The principle of this high-frequency unit can best be likened to a further compacting of the typical cluster of loud-speakers, as customarily used in auditoriums and stadiums for public address systems and announcing, except that the whole array is fed from a common header and driven by two dynamic units. This type of high-frequency radiation is also a feature of the aforementioned reference system. However, the reference horn, having been developed to a very limited angle and being driven by a single mechanism, was not adaptable to theatre use, as more than one horn became necessary for full coverage. This would result in non-uniform distribution as well as complete loss of coverage for a large part of the auditorium, should one unit fail during a performance.

One of the features of the reference system is the use of a single diaphragm to reduce phase distortion. Inasmuch as theatres require parallel operation as protection in the case of failure of one unit, experiments

were made with a Y throat and two units. As a result of these experiments, it is now recognized by all concerned that any increase in phase distortion which may be introduced by the Y throat is negligible.

The diaphragms are made of duraluminum 0.002 inch thick and have an area of six square inches. The diaphragm is mounted on the

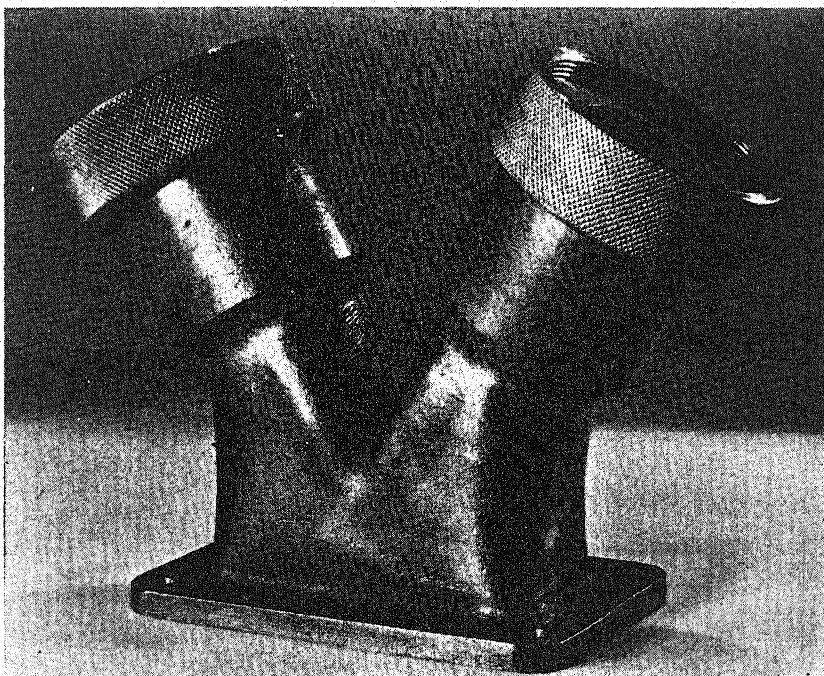


Figure 63 — The Y Throat.

back of the assembly and by the use of an annular opening, the sound that is admitted to the throat within the unit has a minimum phase distortion. (Figure 64.) This is still further reduced by having this throat exponential beginning at the annular opening, and avoids a sharp discontinuity that may exist with a tubular throat. Two units are connected by means of a Y throat to the multi-channel horn which tends to reduce the distortion of high throat pressure. The field excitation requires twenty-five watts per unit.

To obtain high efficiency energy transfer between the diaphragm and air column in an exponential horn loud-speaker, the acoustic impedance of the air must be matched with the mechanical impedance of the diaphragm. Such an impedance match is usually secured by the use of an acoustic transformer which provides a properly constricted

cross-sectional area of the sound channel between the diaphragm and the throat of the horn.

This impedance matching device provides a load for the diaphragm which may be taken as the ratio of the effective diaphragm area to the

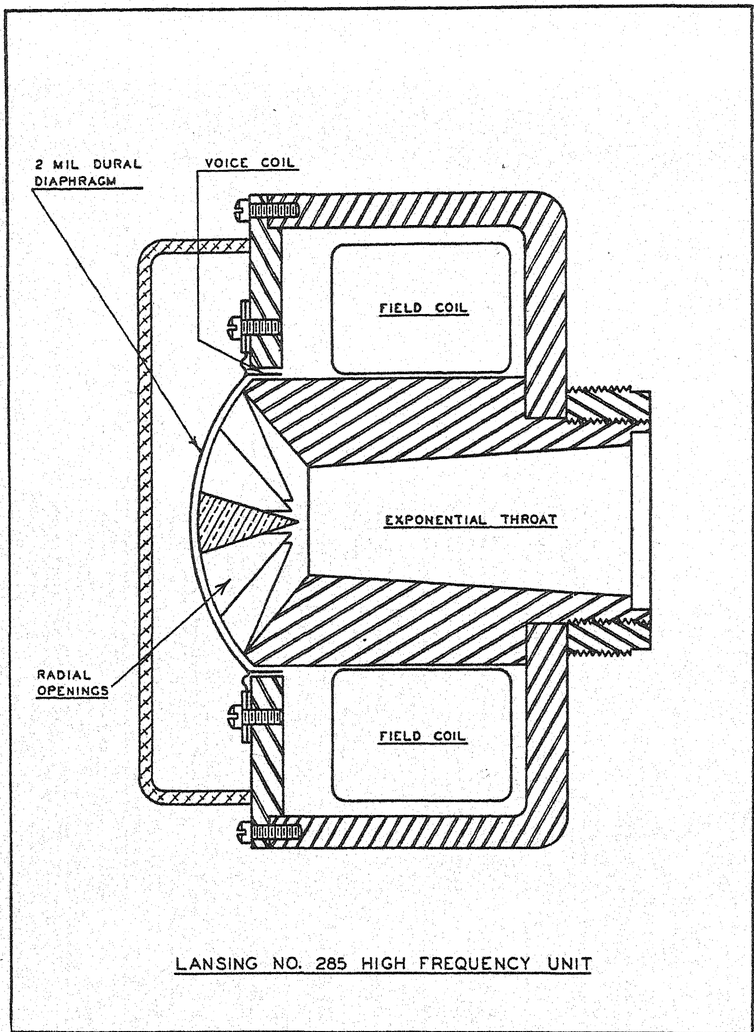


Figure 64.

cross-sectional area of the constricted sound channel. This loading factor, of such a value that it tends to damp out resonant action of the diaphragm, should provide a loading for the diaphragm which is as uniform as possible over its entire area.

The sound channel dimensions between diaphragm and horn must be of such values that waves of any frequency within the recording band which emanate simultaneously from different points of the diaphragm, will not meet at any point in the throat 180° out-of-phase—since in this case destructive interference between such waves would take place.

It is also required that any waves which originate from all points along a radial element or sector of the diaphragm, form a wave front of a desirable shape at the throat of the horn.

One method considered, to avoid destructive interference at the throat between waves which originate at the center and outside edge of the diaphragm, was to provide individual sound channels of substantially equal length from the diaphragm to the throat of the horn. This

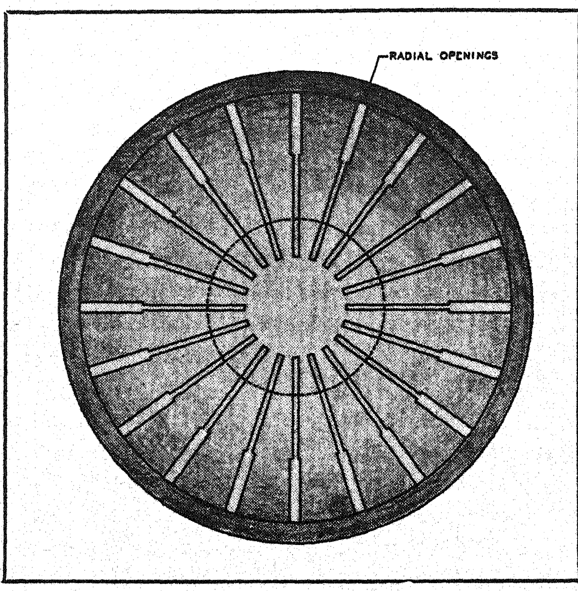


Figure 65 — Slit structure of acoustic transformer—Lansing No. 285, high-frequency unit.

resulted in the construction of acoustic transformers in which there were formed a number of annular concentric sound passages all of substantially equal length.

Such a device proved practical and gave good performance, but was difficult and expensive to construct.

To overcome these practical difficulties, another type of acoustic transformer has been constructed. This type is illustrated in Figure

65, in a view looking into the transformer from the diaphragm. It is rugged, of relatively simple construction, and inexpensive.

The transformer provides sound channels which are disposed radially so that each opening subtends a sector of the diaphragm. Waves originating along a continuously radial sector of the diaphragm will be passed by each sound channel in the manner in which they originate A wave front which emanates from each radial element of the diaphragm advances without further subdivision into the radial sound channel to the throat of the horn, arriving at that plane in an approximately spherical form. No destructive interference is encountered, providing the maximum distance from any point of the diaphragm to the opening of the nearest radial sound channel is less than a quarter wave length of the highest frequency to be reproduced.

A loud-speaker provided with an acoustic transformer designed in accordance with these considerations is shown in Figure 64. Such a loud-speaker reproduces at a response level which is highly uniform throughout the range from 300 to 10,000 cycles.

The directional characteristics of the resulting unit are very satisfactory as found from theatre installations. It should perhaps be emphasized that lack of good distribution cannot be corrected by equalization in the electrical circuits, since for any given adjustment, the overall response is a highly varying function of position in the house. Although the characteristic can be made flat for any given position, it can not be made so for all, or even a large part of the house by this method.

8. LOW-FREQUENCY HORN

In the case of a low-frequency unit, a suitable driving mechanism was not available, and it became necessary to develop one. The unit finally adopted consisted essentially of an exponential horn with a mouth area of fifty square feet, and an axial length of forty inches, driven by four fifteen-inch dynamic units of special design. The mouth opening was extended laterally to form a flat baffle, 10′ x 12′. The paper cones are dipped with lacquer to prevent them from absorbing moisture, which would vary their response. They are connected in series-parallel to give a desirable impedance characteristic as well as to provide insurance against complete failure of the system in the event any individual unit fails. The angle of distribution is uniform through an arc of fifty degrees on each side of the axis. The use of a horn instead of a flat baffle board for low frequencies has several advantages: The efficiency is raised from ten or fifteen per cent to better than fifty per cent, which effects an enormous reduction in amplifier capacity; and undesirable radiation from the rear of the unit is considerably reduced, resulting in a decrease

to a negligible amount of the usual objectionable back stage low-frequency "hang-over." For purposes of further compactness and rigidity the low-frequency horn may advantageously be folded and in this form retains the same characteristic, if the air path length be maintained unchanged. This modification was contributed by Dr. H. F.

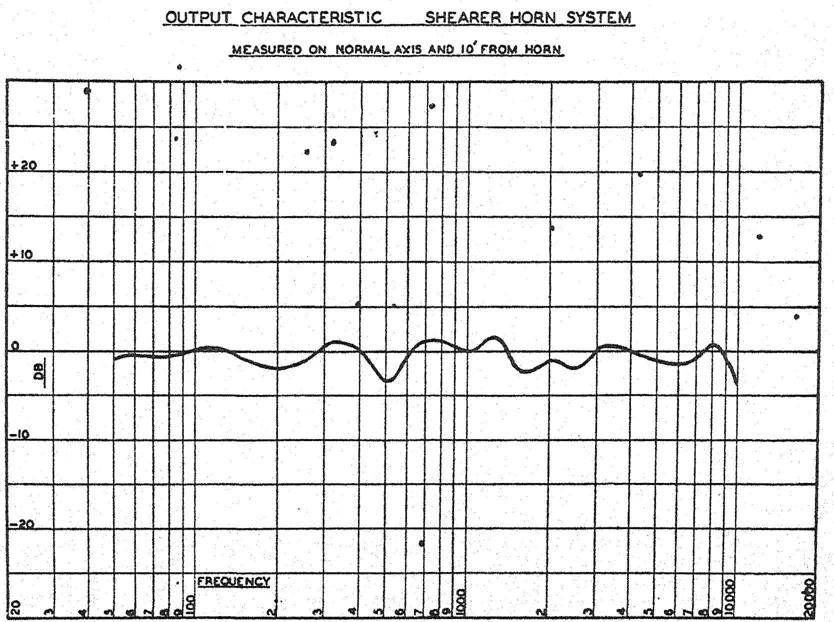


Figure 66.

Olson of the RCA Manufacturing Co. The loading provided by the air column of the horn decreases the excursion of the diaphragms as compared to the excursion necessary to produce equivalent output from a flat baffle array, and distortion is correspondingly reduced. (Figure 66.)

With the low-frequency horn length, as specified in the design under discussion, maintained approximately equivalent to the length of the high-frequency horn, there is no time delay between the component sounds from the two horns.

9. HORN ASSEMBLY

The folded horn is assembled in sections, each section containing two driving mechanisms. They may be stacked one upon the other, depending upon the number required. Each section is adequate for an output from the amplifier of 25-30 watts for the required minimum

harmonic content. If it is desired to secure a wide lateral distribution, the sections may be placed side by side. Section AA, Figure 67, shows the construction of the horn.

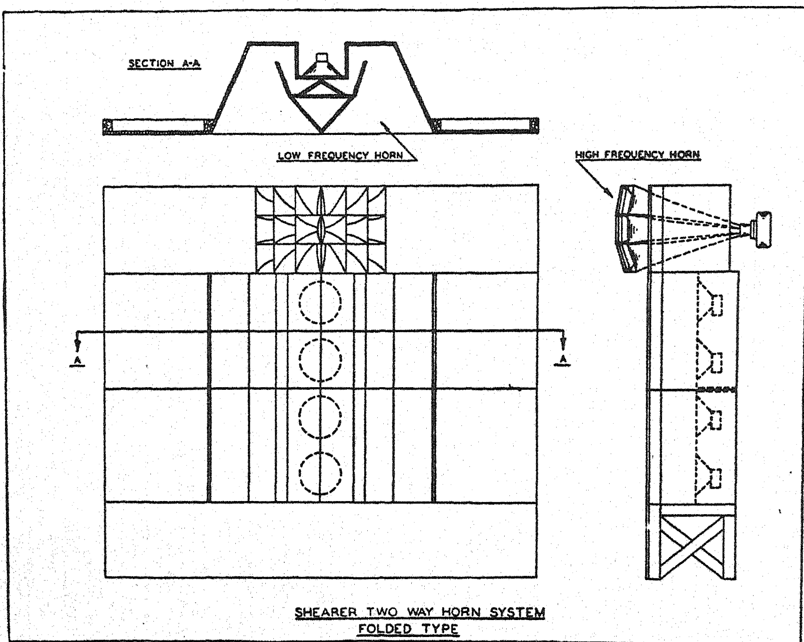


Figure 67.

The entire horn is assembled so that the center of the high-frequency unit is approximately 50 to 60 per cent of screen height. This position has been found by years of use to be the center of activity or "presence" on the screen and since the high-frequencies are responsible for determining the presence, the unit was so arranged. In order to keep the sound as near a point source as possible, the low-frequency horn is maintained at a position near the high-frequency horn. (Figure 68.)

The complete assembly is a unit so that it can be moved away from the screen or raised and lowered with the screen with a minimum of effort. The use of sections for the low-frequency horn allows the horn to be shipped and moved into spaces which have standard size doors.

10. DIRECTIVITY

For both the low- and high-frequency units a certain amount of directivity is required, since the best illusion is obtained if the ratio of direct to reflected sound is as high as possible. In most theatre auditoriums there should be but little energy radiated at angles greater than

about forty-five degrees from the axis, as such energy will be reflected from the walls and will consequently lessen the illusion.

There is one additional consideration with regard to directivity which should be mentioned. Dr. V. O. Knudsen has shown that at

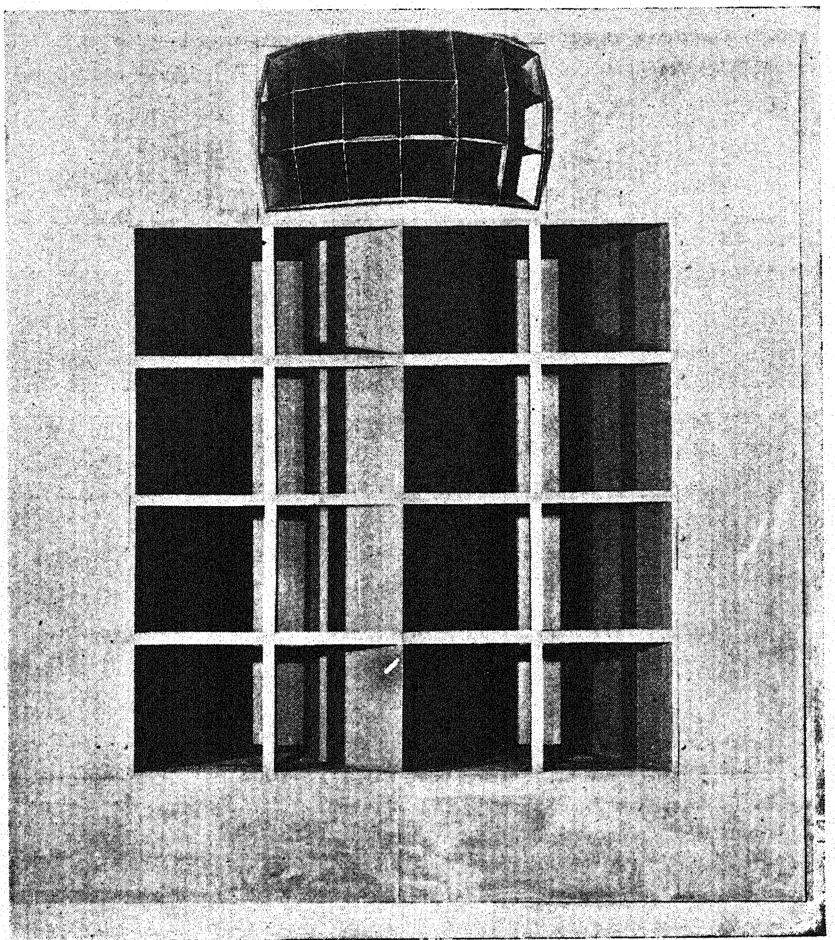


Figure 68 — The folded horn assembly.

the higher frequencies, *e.g.*, at 10,000 cycles, absorption of the atmosphere may become very serious, being as great as 0.2 db per foot under certain conditions of humidity and temperature. In large and deep houses this would result in a serious loss of high-frequencies in the rear seats. The effect can be considerably reduced by increasing the high-frequency radiations from those horns of the unit which serve these seats.

It may be done by putting a suitable amount of absorbing material in the other horns and re-equalizing to bring the overall response up to standard for the front seats. These artifices will probably not be required in most houses.

11. HARMONIC CONSIDERATIONS

A major defect of commercial loud-speakers is their large amplitude distortion. One of the striking improvements in the new system

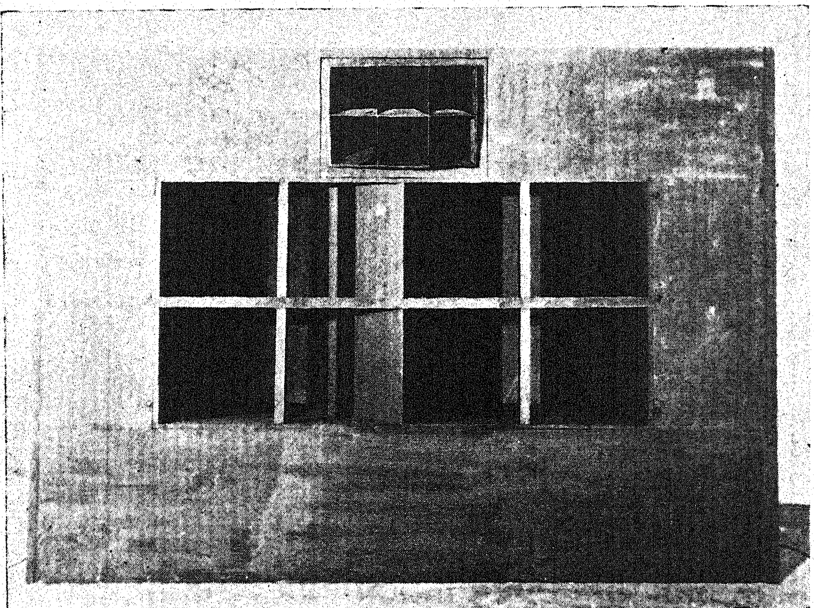


Figure 69 — Single section low-frequency folded horn with a 52 degree high-frequencies for use in studio viewing rooms and small theatres.

is its cleanness of reproduction at low frequencies. The measured harmonic content is less than four per cent at 40 cycles for 30 watts output. This is due in large part to the use of a thick and comparatively soft cone which can be driven to full excursion without break-up, and consequent harmonic production. It was found by actual listening tests that with a pure tone of forty cycles impressed, most of the cone speakers investigated gave a greater apparent loudness than the speaker finally adopted. However, when a direct comparison was made by keying the amplifier from the new unit to the unit under test, it was at once obvious that the output of the new one was fairly pure forty-cycle tone, while that of the other speakers consisted, in most cases, entirely of the second and higher harmonics. Direct measurement of the acoustic

output showed that in spite of its low apparent loudness, the fairly pure output of forty cycles was actually about 6 db higher than that of the other speakers.

This great increase in apparent loudness due to transferring part of the fundamental power into harmonics in the conventional speaker is very striking, and is undoubtedly the explanation for the alleged high efficiency of many present day speakers of all types. The loudness of the harmonics is not due to the rapid change in the sensitivity of the ear at low-frequencies which would favor the harmonics at the expense of the fundamental, since it also occurs at fairly high frequencies where the sensitivity of the ear is varying in the opposite way with frequency. With one particular pair of units tested, the effect was more striking at 1,000 to 2,000 cycles than at any other frequency. It is equally great with complex sounds, such as speech and music, although here the change in quality is somewhat less with respect to the change in apparent loudness than in the case with pure tone.

12. PHASING

Another important advantage of the new system is that it can easily be made to fulfill the requirements that the virtual sources of all the components of the reproduced sound shall coincide in the vertical plane. This condition is impossible to obtain with divided frequency range systems now in use in which the axial length of the several types of horns in a given system are widely different. In this respect, a two-unit system is much easier of adjustment than a three-way system. It might be thought that since the time delay is so small, of the order of a few milliseconds, that the effect would be inappreciable. This is true for certain types of sound such as sustained music passages, but on dialogue and especially certain types of sound effects which are of the nature of short pulses, a very objectionable distortion is usually noticeable. A striking demonstration of this fact was obtained by recording a tap dance. When this was reproduced it was found that the system with a very small time delay gave a naturalness of reproduction, while systems which had an appreciable delay reproduced the scene with far less realism. In fact, the sound did not appear to come from the screen, and, in addition, the tap was fuzzy in character with a decided echo.

This effect sounds somewhat like that of transient distortion due to the use of a filter with too sharp a cut-off, but it is actually more analogous to the echo effect often observed on long lines and with certain types of phase distortion networks.

A recent paper discusses the features of the three-way system, including some of the limitations which require special installation tech-

nique for the setting of horns, back stage draping, phasing of various horn positions, position of horns for distribution and setting of volume between horns. Familiarity with this data will assist in appreciating the principles of the present system.

It should be pointed out that the overall frequency response curve of the system should not fall off too rapidly beyond the cut-off fre-

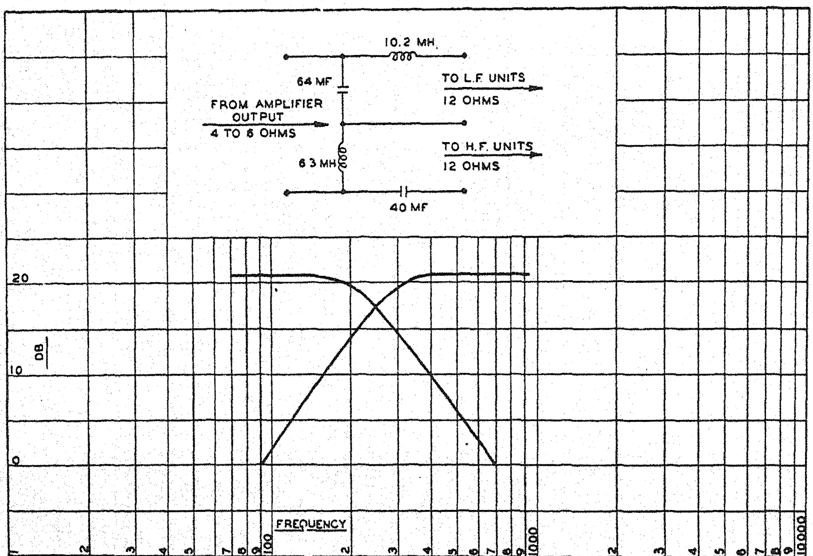


Figure 70 — Series type dividing network, Shearer Horn System.

quencies, or objectionable transient distortion will result. Probably the maximum slope that can be tolerated is of the order of 20 db per octave, or roughly, that of a single section constant-K filter.

13. DIVIDING NETWORK

The frequency chosen for the critical frequency of the dividing network is governed by several factors. If this frequency is too low, it leads to uneconomically large values of capacity in the network, and to impracticably large horns for the high-frequency unit. If too high, there is danger of running into the characteristic dip which seems to be always present in large cones, and also, it would result in dividing the prime energy of speech sounds between the two units, which is objectionable from the standpoint of good presence. If the critical frequency is chosen between 250 and 400 cycles, a good compromise results. (Figure 70.)

A dividing network was chosen which gave fairly rapid attenuation, 12 db per octave, in order to keep any appreciable low-frequency energy out of the high-frequency unit, and to minimize the effect of irregu-

larities encountered in the response curve above the designed range of the low-frequency cones. This lies somewhat above 400 cycles for an efficient low-frequency unit. Certain dividing networks in current use have attenuation curves of such gradual slope that at some frequencies the irregularities in the response curves of the speakers are actually greater than the attenuations of the networks.

The network is designed so that the reflected impedance of the horn on the amplifier is approximately 2.5 times the amplifier impedance. The loss in the network is less than 1 db in order that the full capacity of the amplifier may be utilized.

14. MEASUREMENTS

While it is recognized that indoor response measurements do not have the degree of precision that may be had in free space, they nevertheless do represent conditions under which the loud-speakers must actually be used for motion pictures. Also, for the purpose at hand, comparative measurements are sufficient and were verified by listening tests, which in the end are the final criterion. (Figure 66 shows average response.)

Irregularities in the sound pressure at the microphone due to standing wave patterns in the room are minimized by the use of a conventional warble frequency, varying plus and minus twenty-five cycles at a ten cycle rate. Tests have been run which indicate that the warble is only effective below 2,000 cycles. Above this point, the standing waves do not interfere with the correct interpretation of the response curve.

The measurements were taken in a stage 100′ x 70′ x 35′, having a reverberation time of one second at 512 cycles per second. By making these measurements indoors, tests could be made rapidly on a large number of units without interference from outside noises, due to a 60 db insulation between inside and outside, provided by the building.

The response curves were measured using a high speed level indicator capable of responding to a change in level as rapid as 300 db per second.

Douglas Shearer, head of the Metro-Goldwyn-Mayer Sound Department, brought about and directed this project. This development was engineered by the writer and contributed by Metro-Goldwyn-Mayer Studios. The cooperation of the following companies is gratefully acknowledged: Electrical Research Products, Inc.; RCA Manufacturing Co.; Lansing Manufacturing Co.; and Loew's, Inc. These companies assisted by making available test equipment the reference system, staff and theatres, which greatly facilitated the work and pro-

duced a co-ordinated result not otherwise possible. The writer also wishes to acknowledge the contribution of the Metro-Goldwyn-Mayer Sound Department, and in particular Robert L. Stephens, who has carried out the mechanical design.

15. DESIGN DATA

(a) Low-Frequency Exponential Horn

Fundamentally, the design of a low-frequency exponential horn follows the same treatment as that accorded a horn for high-frequency response. There is, however, a greater tolerance allowable in deviating from theoretically calculated values, namely: Expansion rate (governing value of cut-off frequency), mouth size, and nature of cross-section. Discontinuities which would be out of the question in high-frequency design may be permitted with little loss in a low-frequency horn. Numerous tests have borne out the above statement. A horn of folded cross-section has been chosen for general use in this system, because it permits a compactness of design not possible with a straight exponential horn. Sufficient loading has been obtained in a small space to permit the cone driving units to operate at their optimum efficiency.

For the purposes of illustrating the method of computation, a brief summary of the calculations involved in the design of a straight exponential horn will be given:

The cut-off frequency was chosen at 50 cycles per second. A 50 cycle wave has a length of 271 inches. The distance across the mouth of the horn should be at least equal to one-quarter the wave length of the lowest frequency it is desired to transmit. This value for the horn in question gives a minimum mouth size of 68 inches. The size of throat must be sufficient to accommodate four 15-inch cone speaker units. A throat size of 30 x 30 inches was chosen.

It has been found that an exponential horn whose area doubles every 12 inches will have a cut-off frequency of 64 cycles per second; one whose area doubles every 6 inches, a cut-off frequency of 128 cycles per second. From the above relationship the length for the area of the present horn to double, may be found by simple proportion:

$$\frac{64}{X} = \frac{50}{12}$$

from which $X = 15.36$ inches

From the general horn equation:

$$S_x = S_1 \epsilon^{kx} \qquad \text{(Eq.18)}$$

S_X = cross-sectional area at any point X
$\epsilon = 2.7183$
M = flare constant of horn
X = distance along horn axis from throat

where S_1 was chosen above as 900 square inches, M can be computed by substituting known values in the above equation:

$$1{,}800 = 900 \times 2.7183^{M15.36}$$
$$\text{from which } M = 0.045$$

Then the equation for the present horn becomes:

$$S_X = 900\,\epsilon^{0.045x}$$

from which the sectional area at all points X may be computed.

For a minimum mouth area of 4,624 square inches, the length is determined:

$$S_X = 900\epsilon^{0.045x}$$
$$4{,}624 = 900 \times 2.7183^{0.045x}$$
$$\text{Where } X = 36\tfrac{1}{4} \text{ inches}$$

It has been found, however, that while the sizes above are satisfactory from a theoretical standpoint, an increase in loading will result in a higher efficiency. An increase in length to 44 inches with a corresponding mouth size of 80 inches or 6,400 square inches has, as a result of tests, proven to be perhaps the most desirable size. The overall length, inclusive of units, then becomes approximately 55 inches. This length is considerably more than is desirable for the majority of installations.

The above analysis applies to the straight type horn rather than the folded type.

Figure 67 illustrates a horn of folded cross-section. Here it is possible to retain optimum loading conditions in a minimum space. It is, however, in this case mechanically impracticable to construct a horn of true exponential shape.

The mouth, throat size, and flare constant are determined as in the case of the straight exponential horn. Intermediate cross-sectional areas are approximated to those of a true exponential horn as closely as is feasible without involving constructional difficulties.

It has been found that the difference in response is sufficiently slight to justify this deviation from the theoretical.

(b) High-Frequency Exponential Horn

The specifications require that the overall depth or length of both low- and high-frequency assemblies does not exceed 44 inches.

This limitation of length brought about the selection of a theoretical cut-off frequency of 220 cycles per second. This value of cut-off allowed the design of a horn which fulfilled the desired requirements, such as a spread of either 90 degrees or 105 degrees with a maximum of six separate channels and a sufficient mouth size to present a reasonably small amount of discontinuity.

A brief summary of the design calculations follows:

It has been found that an exponential horn whose area doubles every 12 inches will have a cut-off frequency of 64 cycles per second; one whose area doubles every 6 inches, a cut-off frequency of 128 cycles per second. Then by simple proportion the length for the area of the present horn to double may be found:

$$\frac{64}{X} = \frac{220}{12}$$

from which

$$X = 3.5 \text{ inches}$$

From the general horn equation:

$$S_X = S_1 \epsilon^{Mx} \qquad \text{(Eq. 18)}$$

where

S_1 = area of throat (chosen as ¼ square inch)

M can be computed by substituting known values in the above equation:

$$\tfrac{1}{2} = \tfrac{1}{4} \times 2.7183^{3.5M}$$

from which

$$M = 0.2$$

Then the equation for the present horn becomes:

$$S_X = \tfrac{1}{4}\, \epsilon^{0.2x}$$

From which the sectional area of the horn at all points X may be computed.

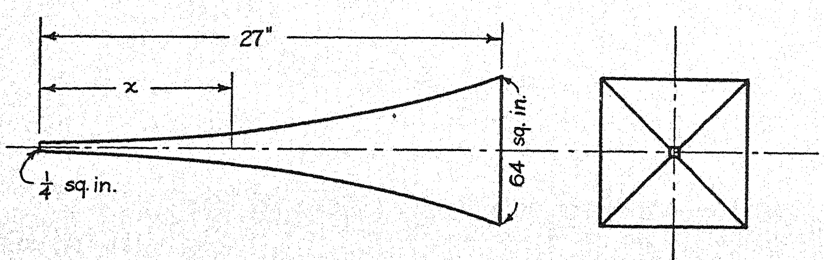


Figure 71 — A typical individual channel.

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Chapter VIII

FILM DRIVE

By WESLEY C. MILLER

Uniformity in the speed of the record and perfect synchronism between sound and picture are vitally important to the sound picture. While the subject matter in this book is largely devoted to the problems of transmission, a brief mention of the film-driving means may be desirable.

The internationally accepted standard film velocity for 35 mm. film is 90 feet per minute or 18 inches per second. Film has 16 frames to the foot, each frame having four sprocket holes so that this velocity corresponds to 24 frames or 96 sprocket holes per second. In the case of a picture, an intermittent motion device in both camera and projector causes each frame to remain stationary for a portion of its twenty-fourth of a second, and then to be pulled down to allow the next frame to take its place. Sound film, on the other hand, moves steadily past the recording or reproducing optical center, so that each three-fourths of an inch (four sprocket holes) of sound track corresponds to the accompanying picture. Inasmuch as eye and ear fail to recognize inaccuracies smaller than perhaps half a frame, the illusion is properly maintained.

In newsreel work, where speed and simplicity of operation are imperative, and where a relatively small amount of complicated editing is done, the sound record and the picture are both made on the same negative film with an offset of several frames between them to allow for the physical separation of picture and sound optical systems in the sound camera.

Two methods are used in the studio for simultaneously driving sound and picture apparatus: The synchronous system and the interlock system.

1. THE SYNCHRONOUS SYSTEM

In the studio the universal practice is to make the picture and sound on separate films which run through separate cameras and recording machines. The practical advantages are many. Negative film emulsion and processing for the best results are not the same for picture as for sound. Greater flexibility of operation of camera and recorder, increased

ability to combine and delete portions of either picture or sound, and generally better possibilities of editing are important considerations.

The use of separate films presents a driving problem which is solved by various means, depending upon the specific conditions. Camera and recorder must be geared together in some manner to maintain synchronism. As mechanical connection is not feasible, electrical means are employed. These are of two general types—synchronous and interlock, with various modifications and refinements in each.

The synchronous drive, as its name implies, employs synchronous motors to drive both camera and recorder, the power being derived from the same source. The source is usually the commercial three-phase alternating-current system or its equivalent so that frequency uniformity is excellent. Synchronous motors which have no hunting characteristics are used. The result is absolute uniformity of motor speed and constant synchronism between sound and picture. Many frequency combinations are used. For example, one of the major studios having a 50-cycle commercial supply uses a 50—60-cycle motor generator to provide 60-cycle power to drive the recorder, which happens to be designed to operate at 1,200 R.P.M., a synchronous speed obtainable from 60 cycles. At the same time, a 50—48-cycle frequency changer is used to provide 48-cycle power to operate 1,440 R.P.M. synchronous motors to drive the cameras. As both of these new frequencies are generated synchronously with the original supply, they are synchronous with each other. Other frequency combinations are used for particular purposes.

An explanation of the choice of 1,440 R.P.M. drive from 48 cycles may be of interest. The camera shutter operates 24 times per second or 1,440 times per minute. Its main shutter shaft consequently operates at a speed of 1,440 R.P.M. A motor driving it at this speed requires no extra gearing to adapt it to the camera. This not only reduces power requirements to some extent, but is extremely valuable in minimizing extraneous noise on the set, which might be occasioned by such gears. Moreover, the 1,440 R.P.M. speed was found to be a good compromise for motor size. The camera silencing bungalow or "blimp" must, of course, be made large enough to completely enclose the camera motor. Hence, size and weight penalties are incurred which are out of proportion to the motor itself, if the latter is made too large. Incidentally, camera motors may be greatly over-rated in their temperature rise, as the greatest possible length of time they may operate continuously is for the length of one one-thousand-foot roll of film—nominally about eleven minutes. There is always a period for cooling while the camera is being reloaded. In practice, a camera

motor is seldom called upon to operate continuously for the full eleven minutes.

With separate film, and with the synchronous system, a means must be provided to properly align the picture and sound track for subsequent running—"start marks" is the term employed for this means. In the simple case, the "hand-clap" serves this purpose. After both machines have attained synchronous speed an operator claps his hands in front of camera and microphone. The camera photographs his hands coming together and the sound film shows the small, easily identified striation group which was made by the sound of the clap. These are satisfactory start marks. In practice, start mark systems run all the way from the simple hand clap to fairly complicated automatic systems.

2. THE INTERLOCK SYSTEM

Another commonly used driving system employs the Selsyn type of motor and is known as the interlock system. This is virtually an electrical gear system, whereby all of the motors connected together on several separated units will start together, come up to speed at the same rate, and continue to run at identical speeds. The master gear or distributor is driven from a synchronous or other constant source so that its speed is accurately maintained. As it is interlocked with the other motor units, uniformity of their speed is assured.

As is the case of the synchronous drive, there are many combinations of distributor and motor speeds in use to meet various requirements. A common case is a 2,400 R.P.M. interlock motor for camera drive—geared down to 1,440 R.P.M. at the camera—and a 1,200 R.P.M. interlock drive for the recorder. Both motor speeds are obtainable from the same distributor by selection of suitable combinations of numbers of poles in the distributor and motor windings. Another arrangement is the use of a single synchronous motor which drives two distributors, one directly at 1,200 R.P.M. for the recording machines, and the other, through gears, at 1,440 R.P.M. which is used to drive a number of re-recording machines at 720 R.P.M.

A very useful application of the interlock system is in projection background process work. Here, camera shutter and process projection shutter must be accurately interlocked so that the camera exposure bears a definite relation to the time of projection of each frame. This relation is maintained by an interlock motor system which is again tied to the recording machine either through the same interlock distributor or through some form of synchronous drive.

Interlock drive is very useful for re-recording work. Here several film records must be reproduced simultaneously and accurate syn-

chronous relations must be maintained between these sound tracks, the picture which accompanies them, and the recording machine which records their combined output. It will readily be seen that the interlock drive provides a reliable and facile means of bringing all the machines up to speed together and of maintaining their speed relation throughout the duration of the record.

An interesting variation of the interlock system has proven very valuable for location work where light weight apparatus is required. The motors used for recorder and camera are virtually a specialized form of inverted rotary converter. Battery power drives the motors, while three-phase alternating-current connections to their windings produce a low voltage alternating current which serves to interlock the several units together. The battery sources may be separated—one to each motor unit, in which case one unit is used as the master speed control and the others are individually adjusted to it. A more satisfactory arrangement is a central battery supply, usually at the recording machine, which furnishes direct current to all of the motors. Field rheostats for each motor are located in one place and all speeds are adjusted to maintain interlock at the estimated camera and recorder loads. Final speed control and provision for starting in the interlock condition are maintained by the central operator—the recorder operator. This method of drive is becoming more and more popular for portable use to the extent that it will probably be the best ultimate means to be found for this purpose.

3. THEATRE MOTOR DRIVES

Theatre motor drives are in general of two types. One type employs a motor of peculiar electrical design such that its speed is maintained constant within narrow limits by means of electrical speed control networks. This driving means is widely used but it appears to be in the process of being superseded by another simpler method.

The newer apparatus being installed uses, for alternating-current installations, a single-phase induction motor having unusually small slip. The slip of this motor is determined for the particular projection machine load and the connecting gearing is so designed that with normal motor slip the projector drive very closely approximates synchronous speed. In other words, if synchronous speed requires a motor speed of 3,000 R.P.M. and the motor slip reduces the motor speed to say 2,900 R.P.M., then the motor will be connected to the projector through gearing which has a speed increase of the ratio of 3,000 to 2,900 built into it.

This particular arrangement has been worked out to accommodate the large majority of houses where three-phase supply is not readily

available, and is quite satisfactory. In direct-current installations, the usual practice is to provide a motor generator set to generate an alternating-current supply, and then to use the previously mentioned single-phase induction motor.

4. UNIFORMITY OF FILM MOTION

Motor drive for the film-running device is but one part of the film motion problem and is perhaps the simpler part. Given a motor running at uniform speed, it is required to couple this motor to a mechanism which will run the film itself at constant speed. As the ear is very sensitive to the results of speed variation in either the recording or the reproducing process, the uniformity of the film motion must be essentially perfect. A brief mention of some of the limitations and some of the devices used may be of value.

To begin with, the celluloid which forms the base for the film is a material which is susceptible to fairly wide variations in length due to changes in its moisture content. Its shrinkage is a known element at the various stages of the several processes, but unfortunately the limits within which shrinkage may be maintained are of relatively considerable size. Because of shrinkage variations it is imperative that the sprocket holes be used at some stage of the film pulling process to maintain synchronous speed and to control the loops which form a part of any film driving and pulling mechanism. For example, a one per cent shrinkage may readily occur and this means a change of 10 feet in length in a 1,000-foot reel. Picture mechanism definitely requires sprocket drive to ensure that the four sprocket holes corresponding to each frame are constantly pulled down by the intermittent mechanism to keep the right frame relation. In projection there is a constant 20-frame difference between picture and sound—the picture and sound apertures are just that distance apart—and this difference must be maintained throughout the whole reel to keep in synchronism.

The conventional sound film drive uses the sprocket holes to do the main film pulling and to maintain isolating loop relations, and then departs from the sprocket pull at or near the sound aperture. A roller drum coupled to a flywheel, or some similar device, maintains a high degree of approximation to absolutely uniform motion at the actual aperture. The transition between sprocket hole pull with shrinkage variation and the continuous motion at the drum, occurs in the isolating loops before and after the drum.

The design and operation problems, then, are those of insuring perfectly uniform drum action, of making the film move absolutely with the drum at the instant the former crosses the optical center, and of iso-

lating the film on the drum from any reflected sprocket tooth action from the sprockets immediately ahead of or behind the drum. The failures in these respects are noticeable in two general types of speed variation or flutter. Slow variations in film speed occasioned by motor speed changes, or more often by drum speed irregularities, appear as "wow-wows," to use the vernacular of the trade. High speed variations, which generally bear a definite relation to the 96 sprocket hole per second speed of the film, result in a broken up and harsh character in the record. These variations appearing in either the recording or reproducing process, are bad. If they occur in both processes, it is apparent that the reproduced sound will be very unpleasantly distorted.

In the early days of sound, flutter of the various kinds was a disturbing factor which was apt to occur at any moment and could not be controlled. A great deal of effort has been devoted to remedies, and present day recorder and reproducer design has minimized the possibility of these troubles occurring. Studio recorders have been commercially free from flutter for a long time. Unfortunately, the same may not be said for the many theatre applications of the older apparatus designs which are still in use. It is the hope that speed variation difficulties will be reduced to the point of being inappreciable as the apparatus now available is gradually used to replace the older installations.

Chapter IX

FILM PROCESSING

By L. E. CLARK

In sound recording, film, while acting in the nature of a delay circuit or storage device for the electrical energy of the signal, also controls the relative instantaneous amount of light reaching the photocell in the reproducing mechanism.

As previously explained, the same principles apply to the variable area and variable density methods of recording, but in the two systems different characteristics of the film are employed. It is the purpose of this chapter to explain briefly the chemistry of film processing and the application of the science of sensitometry to the commercial production of high quality sound recording.

Sensitometry, as its name implies, is very largely concerned with the measurement of sensitivity. In its modern applications, however, it embraces a much wider field and may be more completely defined as the qualitative measurement of the response of photographic material when exposed to light or other forms of radiant energy.

Motion picture sound recording on film must go through the following steps before the production is completed and ready to be shown in the theatre: The picture and sound tracks are each recorded separately on different negatives, which are then developed, and the positive, containing both the picture and the sound track, is then printed from these two negatives—this positive, called a composite print, is then developed for use in the theatre.

1. FUNDAMENTAL MEASUREMENTS

The fundamental film measurement is "transmission"; that is, the percentage of incident light which is transmitted through the film, and is expressed mathematically by

$$T \text{ (Transmission)} = 100 \times \frac{L_1}{L_0} \quad \text{(per cent)} \qquad (20)$$

where

L_0 = total incident light falling on the film and
L_1 = total amount of light passing through the film.

It can be seen that transmission, when expressed in per cent, varies from a maximum value of 100 to a minimum value of zero.

Although the measurement of this ratio is the fundamental measurement made on film, it is not usually expressed in this way but in terms of the "opacity," which is the reciprocal of the transmission, and is expressed mathematically as:

$$D \text{ (Density)} = \log O \text{ (Opacity)} = \log \frac{1}{T} \qquad (21)$$

From this it may be seen that as the transmission, expressed as a decimal, varies from one to zero, the opacity will vary from one to infinity and the density from zero to infinity.

Density is the common term used to express the measure of transmission or opacity of film.

2. DENSITY MEASURING DEVICES

Several instruments are available for measuring the ratio of the incident to transmitted light, expressed in terms of the logarithm of opacity or density. These instruments are called densitometers and are divided into one of two types, either optical or electrical, depending upon the manner in which the measurement of density is made.

(a) Electrical Densitometers

This type contains a light-sensitive cell and measures first the light without film in its path and then the light after passing through the film. The ratio of the two values is a measure of the transmission.

This type is designed to employ either a steady or an interrupted source of light depending upon the purpose for which it is intended.

(b) Optical Densitometers

This type is the most commonly used and is either polarizing or non-polarizing; the polarizing type consisting essentially of a polarizing photometer head with a suitable source of light to obtain the necessary illumination, and the non-polarizing type containing no polarizing prism.

Figure 72 shows the main constructional details of the polarizing type densitometer. S and S' are apertures through which the two light beams enter, and after being polar-

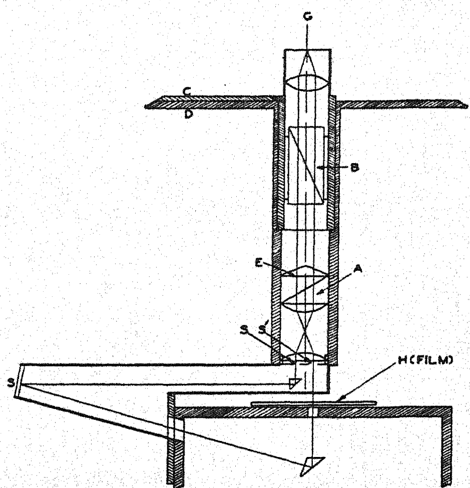


Figure 72 — Details of a polarizing type densitometer.

ized by means of the Prism A, result in two light beams whose planes of polarization are perpendicular to one another. Each beam is then directed through the analyzing prism, B, which is held in a rotating support inside the frame of the photometer head. The member D of the frame and the member C of the analyzing prism support, form a scale which serves as a means of measuring the rotation of the prism B with respect to the fixed prism A. The light field, as seen by the eye from position G, has been divided along a diameter by an image of the apex of the bi-prism E, and thus consists of two semi-circular light fields whose relative intensity is determined by the amount of rotation of B. If both light beams are uninterrupted, prism B would be rotated 45° from prism A to secure light fields of equal intensity.

If the material, the transmission of which is to be measured, is placed at H, then the beam passing through this side of the photometer head will be changed in intensity and further rotation of the prism B will be necessary before the two light fields again have the same intensity. From the new position of the scale C, the density may be directly computed.

The non-polarizing type is illustrated in Figure 73, and consists of a movable source of illumination A; two diffusing screens, B and C; four reflecting mirrors D, E, F and G; and a lens H. The photometer head is shown in detail in Figure 73-B.

The illumination on the screens B and C varies as the light source moves, and is inversely proportional to the square of the distance from

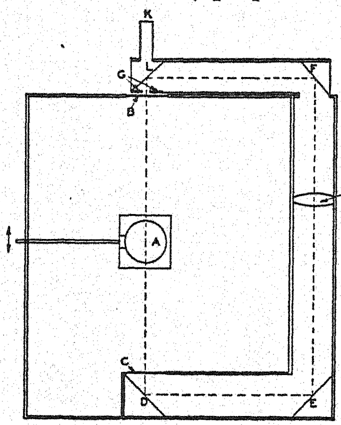


Figure 73-A — Non-polarizing densitometer.

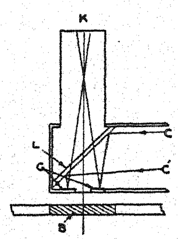


Figure 73-B — Details of photometer head used with a non-polarizing densitometer.

source to screen. As A moves toward B the intensity on B increases, according to the inverse square law, just as it decreases on C. Images of screens B and C are formed in the photometer head as shown. L is a

pane of clear glass placed at an angle of 45°, which serves to both reflect and transmit the light rays. With no absorbing material placed in the instrument, the light source A would be at some zero position depending upon the losses of light intensity in the various mirrors and lenses.

The material to be measured is then placed between the screen B and the reflecting mirror G, so that the beam passing through the screen B also passes through this material, and the eye sees a light field consisting of that part which passes through the material to be measured, surrounded by a field which comes directly from the light source. By moving A to a position where these fields are equal, the inverse square law may be applied to compute the density. A suitable scale attached to A permits the reading of these values directly.

There are, of course, many different kinds of densitometers available of the above types, each most suitable for particular uses.

3. DENSITY

Motion picture film is composed of a celluloid base approximately five-thousandths of an inch thick, carrying a thin coating of emulsion, the principal contents of which are gelatin, silver bromide and silver iodine. When the film is exposed and developed, the silver bromide, under the action of the light rays and developing solution, is converted into a layer of opaque silver, rendering film less capable of transmitting light than before exposure and development.

The maximum density obtainable depends upon the amount of silver present in the emulsion. In the case of sound positive this maximum density is usually from 3.5 to 4, but densities of this value are never required in commercial work.

4. DEVELOPMENT

If a piece of film is exposed in the light for a short period of time there will be no discernible difference in its appearance until it is placed in a proper developing solution where it will react with the solution and metallic silver will be formed on the celluloid base. To obtain this deposit of metallic silver the solution must contain a reducing agent, which is usually some organic substance which reacts with the silver bromide to form metallic silver. The reducing agent is used up in the process—and is usually a coal tar derivative broken down in certain definite ways (trade names Metol, Elon, etc.).

If the developing solution contained nothing but the reducing agent the process would be slow, as considerable time would be required for the solution to penetrate even the thin layer of gelatin and to react with the silver. For this reason the solution also contains an alkali, called an accelerator, as its purpose is to speed up the reaction. This alkali may

be sodium carbonate, caustic soda, etc. The speed of the action desired depends upon the strength of the accelerator used, and determines the graininess of the developed film.

The next substance needed is a preservative for the developer, as the solution absorbs oxygen from the air and spoils in a very short time if left standing in contact with air. This preservative is usually sodium sulphate.

One other substance, usually potassium bromide, is added which acts as a retarder or restrainer, the quantity necessary depending upon the type developer required for the particular type of film being processed.

The active agent in this developing solution is the reducing agent. All the others are added to either prolong the life of the solution or to determine the speed of its action as a developer of the film. As previously brought out, all of the silver bromide is not acted upon and it is necessary to remove this residue after developing, so the film is washed thoroughly in water and then put through a so-called fixing solution after it leaves the developing solution. This fixing solution contains "hypo" (sodium thiosulphate), its only purpose being to dissolve the silver bromide left in the gelatin.

In some cases a hardening agent is added at this point, the function of which is to harden the gelatin and to protect it against scratches, cuts, etc. After again being washed and dried, the film theoretically consists of clear cellulose covered by particles of silver, the amount of which determines the density.

5. SENSITOMETERS

Obviously, the type of solution, type of film, and amount of exposure for a given result have not yet been determined, and it can be seen that it would be of great value to be able to predetermine these points and to control results after these variables are decided upon. This is accomplished by the use of sensitometry in film processing.

In exposing, developing, and reproducing a film record, the type of film, the developing solution, the exposure, the time of development, printing, etc., are all variables, and unless carefully controlled are apt to give confusing results.

The effect of exposure on a given film which is to be developed in a standard solution for different periods of time will first be considered.

Instruments for exposing photographic material for a series of graduating and precisely known periods are known as sensitometers, and consist of a light source of known intensity and a means of produc-

ing different exposures of known relative amounts. The light source affects the results both by its intensity and spectral composition.

The exposure modulator may be one of several types, as the exposure may be varied by controlling either the intensity of light or time of exposure. One type uses a rotating sector-wheel allowing the light to pass for a smaller period of time as the distance from the axis increases; another type uses tablets of different density between the light and photographic material, thus changing in a regular manner the intensity of exposure. There are a great many varieties of these instruments, but the general operation is much the same, variations being only in the application of the following principle:

A strip of film, divided into sections, is placed in the field of illumination of a dim light of proper value. At periodic intervals of one, two, four, eight, etc., seconds, the sections are covered by moving an opaque object parallel to the length of the film. At the end of the exposure time each section has been exposed a different length of time, namely: One, two, four, eight, etc., seconds depending upon the number of times this process is repeated.

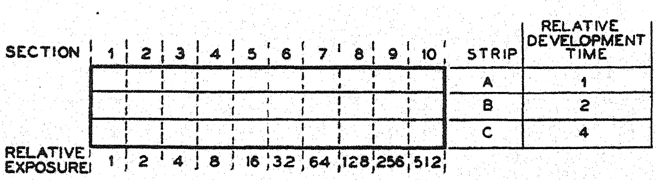


Figure 74.

This film is then cut into three lateral strips and the first developed for two minutes, the second for four minutes, and the third for eight minutes, to give a development time ratio of 1, 2, and 4. After each strip is developed, the density of each section of each strip is then read on a densitometer. Figure 74 shows the three strips of film and the relative exposure and development time for each section.

6. "H AND D" CURVES

Considering each strip separately, the densities can now be plotted with respect to the exposure, and give a curve of the form shown in Figure 75, when the exposure is plotted logarithmically. The usual method is to plot density and log of the exposure as shown by the middle horizontal scale. These curves are known as "H and D" or "density-log E" curves.

It will be noticed that each curve consists of three distinct parts, a toe, a straight line portion, and a shoulder. The toe, region 1-2, is called the region of under-exposure, as changes in exposure have little effect on density. The region 3-4, the shoulder of the curve, is the region of over-exposure, as again changes in exposure cause only small changes in density. The straight-line portion of the curve is called the region of correct exposure as in this region the density is directly proportional to the logarithm of the exposure.

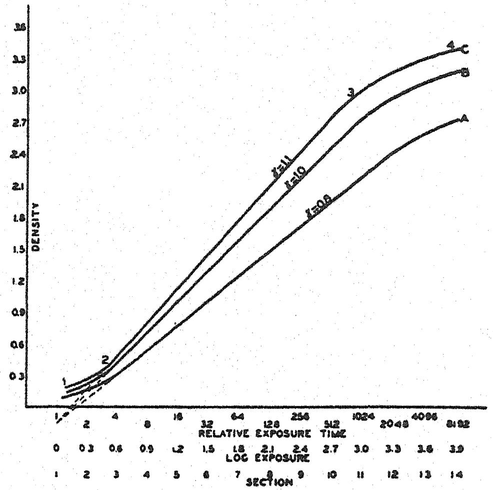


Figure 75 — Curves showing the relation between density and exposure of three strips of the same film at three different development times.

7. GAMMA

From these curves it is possible to predict results from a given film developed for a certain length of time in a certain developing solution. With these factors constant, and considering only the straight-line portion of the H and D curve, the density varies with the exposure in a regular manner. This variation, or slope of the curve, is known as the gamma (γ) of this particular curve, that is, the tangent of the angle between the H and D curve and the exposure axis is called gamma. Thus for the three curves in Figure 75, we have γ_1, γ_2, and γ_3 as shown. Gamma is a measure of the effect of relative exposures on any particular film which is left in a certain developing solution for a given length of time.

If we consider Figure 75 as H and D curves of negative film, it is now necessary to make prints from these negatives which will give another series of H and D curves, depending upon the solution used and length of developing time. Now, if a series of prints were made, each being developed a different length of time, the density of each section of each print would again vary with exposure and an H and D curve for each print could be plotted. It can thus be seen that these conditions, unless controlled, would give rise to varied and unpredictable results.

The relation between the length of developing time and the gamma is controlled by means of curves drawn for each type of film developed

in the same solution. Typical curves showing this relation are shown in Figure 76, Curve (1).

8. FOG

Film subjected to no exposure, but developed for any length of time, would theoretically have a density of zero. Actually, however, it has a definite density depending upon the photographic material used and the development time. As a consequence, the theoretically transparent portions of sound track actually have a certain density, which is called "fog." This is illustrated in Figure 76, where the curves marked (2) are the fog curves of the materials whose gamma-development time curves are also given in the figure.

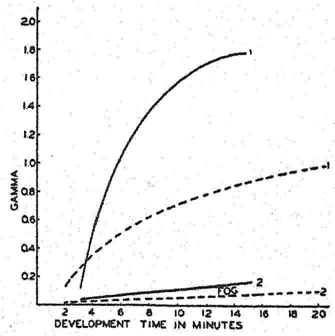


Figure 76 — Curves showing the relation between gamma and development time.

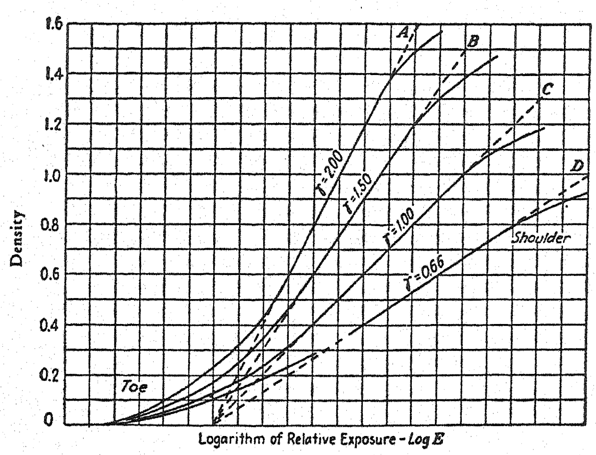
9. PHOTOGRAPHIC REQUIREMENTS OF VARIABLE AREA AND VARIABLE DENSITY RECORDS

It will be remembered that the distinction between variable area and variable density recordings is that in the former both factors of exposure, intensity and time of illumination, are constant at any point within the exposed area, while in the latter method the intensity factor is constant while the time factor varies (due to the changing width of the light-valve slit). This difference in the two types of records necessitates the use of different film characteristics for each type, even though both systems function under the same criterion, namely, that the sound record positive, printed from the negative, shall carry a distribution of density such that the intensity of the light reaching the photocell in the reproducer is directly proportional to the instantaneous pressure on the microphone diaphragm.

(a) Variable Density Requirements

From a photographic point of view, H and D curves are used in the study of emulsion characteristics and the effect of development. However, in sound track processing, where the relation between the input to the film and the output from the film are of primary importance, the overall result is considered. The basis of this relationship is still the H and D curves of the negative and positive films, but as the input of the film is measured by the negative exposure and as the positive trans-

mission is proportional to the output from the film, the most logical method of studying this relationship is between the negative exposure and the positive transmission.



—From "Recording Sound for Motion Pictures," Academy of Motion Picture Arts and Sciences (McGraw-Hill Book Co., Inc.).

Figure 77 — Typical H and D curves.

Typical H and D curves, with four different values of gamma, are given in Figure 77. The equation of the *straight-line portion* of these curves is:

$$D = \log \frac{1}{T} = \gamma \ (\log E - \log i) \tag{22}$$

where $\log i$ = the value of $\log E$ at the intersection of the straight line portion extended, with the $\log E$ axis

i = inertia of the film

Since the quantity of $\log i$ of equation (22) is a constant for a given case, then

$$T \infty E^{-\gamma} \tag{23}$$

for both positive and negative film. Then if N and P as subscripts denote negative and positive, respectively,

$$T_N \infty E_N^{-\gamma_N}$$

and (24)

$$T_P \infty E_F^{-\gamma_P}$$

In printing, the exposure of the positive is controlled by the transmission of the negative so that

$$E_P \propto T_N \tag{25}$$

From this

$$T_P \propto E_N^{\gamma_N \gamma_P} = K\, E_N^{\gamma_N \gamma_P} \tag{26}$$

where K is a constant related principally to the inertia and the amount of light used in printing.

From equation (26) it can be seen that when the product, $\gamma_N \gamma_P$, equals unity, the positive transmission is directly proportional to the negative exposure. When this condition is fulfilled, provided the straight-line portion of the H and D curves are not exceeded, the true transmission ratio for the film is realized.

In practical applications in sound recording, the gamma of the exposure device (light valve), the gamma of the printing machine, and the gamma of the photo-electric cell, as well as the gamma product of the two films, affect the transmission ratio. For this reason, in practice the overall gamma varies from unity, depending upon laboratory conditions and recording equipment.

Typical examples of the various gammas are: gamma of the light valve, 0.96; gamma of the negative, 0.40; gamma of the printing machine, 0.90; gamma of the positive film, 2.00; and gamma of the reproducing equipment, 1.45.

These typical values would lead to an overall gamma of about one.

Practical experience has led to the adoption of the following standards for certain factors:

1. A fine-grained, long-scale emulsion, such as standard positive stock, seems suitable for both negative and positive sound records.

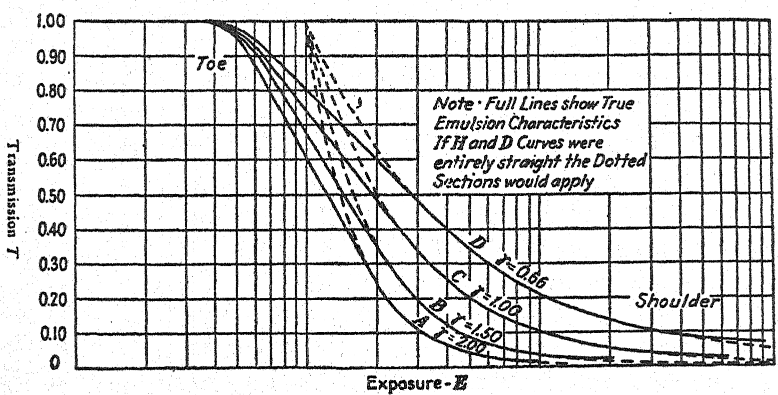
2. Positive development, and hence positive gamma are determined by the picture requirements of the positive. The reason for this is obvious, since both picture and sound track must be developed simultaneously. Commercial positive gamma is usually in the neighborhood of 2.00.

3. Negative development, that is, negative gamma, must be controlled to meet the overall requirements with fixed positive conditions. Negative gamma accordingly ranges from about 0.38 to 0.43.

4. The average positive transmission and the total range of positive transmission modulation must be such that neither excessive light nor excessive amplification is required in theatre reproduction. Accordingly, the average or unmodulated transmission of the positive is

about 0.30. (All statements of density or transmission are made with clear film as a standard, i.e., its density is taken as 0 and its transmission as 1.00.)

5. Negative exposure is theoretically so fixed that the maximum possible value is slightly below the shoulder of the H and D curve. This exposure produces an unmodulated or average negative transmission of about 0.25, depending upon the development chosen and upon practical departures from the theoretical exposure range.



—From "Recording Sound for Motion Pictures," Academy of Motion Picture Arts and Sciences (McGraw-Hill Book Co., Inc.).

Figure 78 — Typical transmission-exposure curves.

To determine the effect of a departure from an overall gamma of unity, and the practical limitations of the above, the H and D curves are drawn in a different form as follows:

The relations between density and log E as given in Figure 77, are replotted as shown in Figure 78, with transmission and exposure as the two new relative values. This may be done by use of the formula

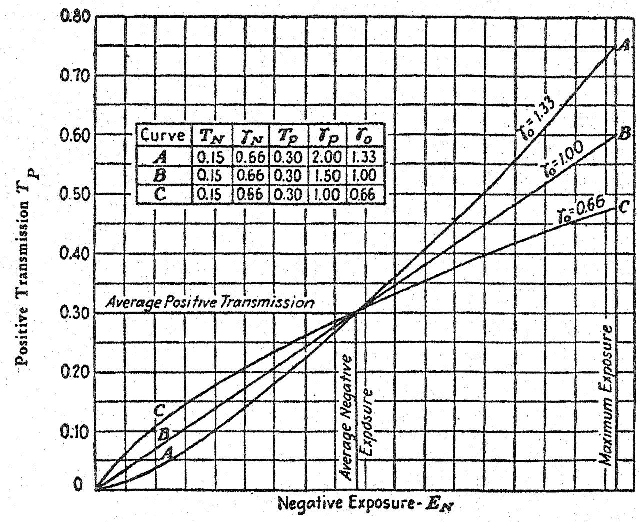
$$D = \log O = \log \frac{1}{T}$$

In Figure 78, it can be seen that the straight portions of the curves of Figure 77 tend to become curved and the curved portions straight, and that the relation between exposure and transmission is not constant.

From Figure 78, and assuming the entire H and D curve to be straight, the curves of Figure 79 can be constructed, using the values of positive and negative transmission, and the combination of gammas, as listed in the figure. These curves show that there is a definite departure from proportionality when the overall gamma is not equal to unity, even when the H and D curve is assumed to be entirely straight.

It is also evident, however, that small departures from unity can be tolerated without a severe change in quality.

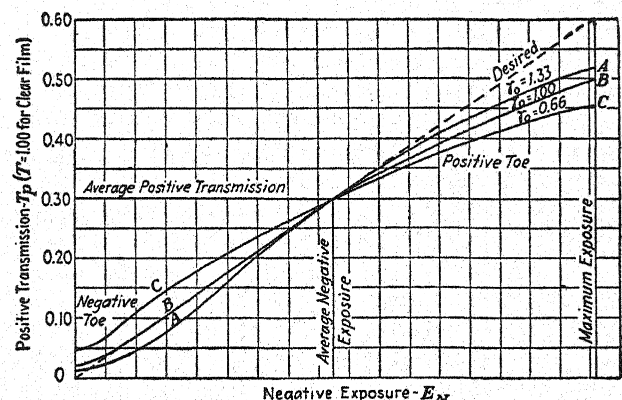
Now if the curves of Figure 79 are redrawn but with true emul-



—From "Recording Sound for Motion Pictures," Academy of Motion Picture Arts and Sciences (McGraw-Hill Book Co., Inc.).

Figure 79 — Curves derived from data in Figure 78.

sion characteristics, that is, considering the curved toe and shoulder portion of the H and D curves, Figure 80 results. This figure shows



—From "Recording Sound for Motion Pictures," Academy of Motion Picture Arts and Sciences (McGraw-Hill Book Co., Inc.).

Figure 80 — Overall relation between negative exposure and positive transmission for various overall gammas showing the effect of toe sections.

that there is a small departure from the theoretical curve in the region of low negative exposures—the result of the negative toe—and large

departures at the high-exposure end—the result of the positive toe.

Considering negative exposure in terms of modulation about the average value, with unity overall gamma, fairly high modulation without distortion is permissible on the low-exposure end, whereas on the high-exposure end distortion commences at practically zero modulation. For overall gammas higher or lower than unity, the extent of this distortion varies. It is a fact, not immediately evident from the figure, that the relative distortion becomes less as the average positive transmission is made less.

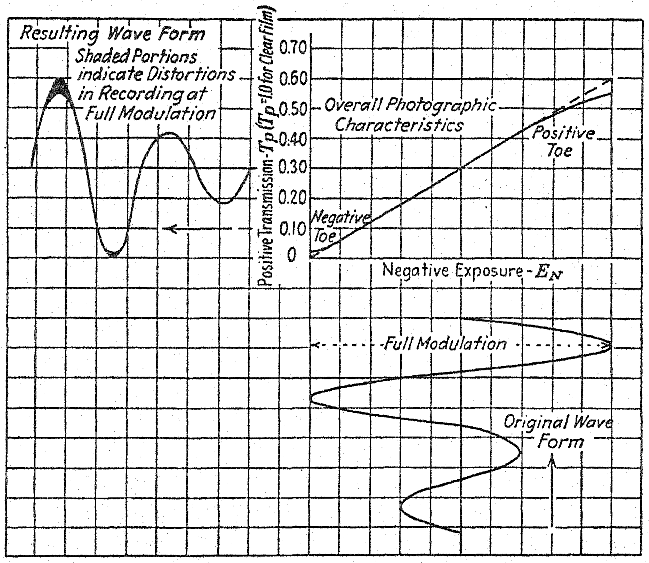
The effect of these toe sections is serious and difficult to correct. Indeed, complete correction is impossible. For low-modulation values they may be accepted, but as commercial records require maximum modulation to keep above surface noise and to maintain proper volume levels, it is often necessary to work right up to the limit of acceptable distortion—and sometimes, perhaps, beyond. The negative toe distortion is under better control than that of the positive, as the amount of use which it receives is a direct function of the control of the exposure device. The positive toe, on the other hand, is the principal source of distortion, and at the same time, is more subject to harmful variation. It seems, unfortunately, that almost every factor in the positive development process is such that it tends to exaggerate rather than to help the positive toe condition. Moreover, production negative can always be more carefully handled than the released positive, as the latter amounts to many more feet than the former, and its production must be undertaken at a reasonable cost.

Referring again to the curves of Figures 79 and 80, there is available an approximate solution to the toe distortion. Experimentally it has been found to be practicable to make relative adjustments of gamma and transmission values, principally the former, to offset one type of distortion by means of another. This fact was not contemplated in the original discussions of the theory of this method of recording.

That such adjustment is successful may be seen from the curve of Figure 81, which is representative of normal results obtained in practice. The major producing studios and the equipment companies employing the variable density method of recording, and working independently or in conjunction with one another, have arrived at this compromise and in every case the results are remarkably close.

It has been arrived at by carefully balancing the several factors involved in the annual production of several hundred million feet of annual release print and represents the best practicable compromise between the commercial and theoretical aspects. As a matter of interest, the same figure shows the effects of film-characteristic distortion upon the repro-

duction of a sine-wave negative exposure. The reproduction is excellent except at high modulation.



—*From "Recording Sound for Motion Pictures," Academy of Motion Picture Arts and Sciences (McGraw-Hill Book Co., Inc.).*

Figure 81 — Results in practice from relative adjustments of gammas and transmission values.

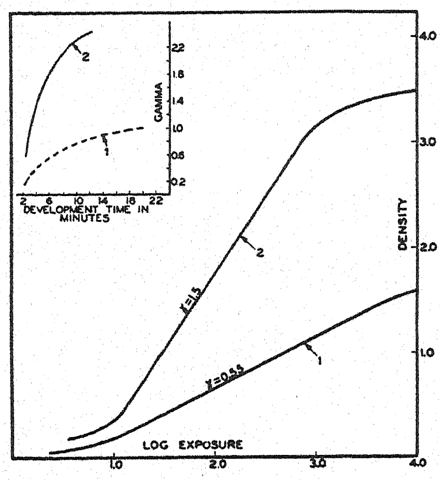
The effect on the mean transmission or no-signal density of the print on noise reduction is considered in the chapter on Noise Reduction.

(b) Variable Area Requirements

With this type of recording, the general requirements of the photographic material are different from those of variable density as the track consists of transparent and opaque areas rather than striations of varying density. For variable area, high contrast between the two regions is desirable, that is, the opaque region should be as dense as possible in contrast to the opposing region, which should be as transparent as possible, with the dividing line between the two very sharp. This calls for a development just the reverse of the variable density method.

This high contrast requirement needs an H and D curve of much sharper slope, that is, with a higher gamma, although the absolute value of this gamma is relatively unimportant. This steeper slope is

necessary in order to obtain the density contrast and image sharpness which are required for this type of track. Also, the gamma-development time curve is much shorter and higher than for variable density recording. Figure 82 gives a comparison of H and D curves and gamma-development time curves for the two types of recording.



Curves (1) for variable density.
Curves (2) for variable area.

Figure 82 — H and D curves with associated gamma-development time curves.

As previously mentioned, the dividing line between the transparent and opaque regions should be sharp, and theoretically at this boundary line there is a region of light and no light, but actually there is always stray light spilling over into the transparent region and resulting in regions of light and less light, which of course reduces the contrast.

This builds up, especially in the case of a high-frequency wave, a sort of halo which follows the wave form and produces a region of graduated density along the wave. This tends to fill up the valleys of the waves and to broaden the peaks in a manner similar to that described under "Slit-Effect" in Chapter III, and distortion and loss of high-frequency results.

To minimize this effect the recording light beam should be focused as sharply as possible.

The new ultra-violet recording, which uses a filter in the optical system to allow only the ultra-violet light to strike the film, has materially reduced this effect by keeping the exposure near the film surface.

The shape of the toe of the H and D curves for both positive and negative are highly important. The transparent portion of the print should have as low a density as possible, in contrast to a density of around 1.5 for the opaque portion. This means that the contrast on the negative must be good, and that the transparent part be of very small density, and, as a consequence, the toe of the H and D curve must be

sharp so that changes in exposure in the lower-density region cause very little change in density.

A comparison between the critical factors governing film characteristics of the two systems shows that for variable density, gamma and density control are most important, while for variable area the shape of the toe of the H and D curve and the contrast are most important.

The effect of noise reduction on variable area processing is discussed in the chapter on "Noise Reduction."

Chapter X

REPRODUCING SYSTEMS

By L. E. CLARK *and* JOHN K. HILLIARD

As brought out previously in this text, release prints for use in the theatre have both picture and sound on the same film, and the reproducing system consists of two more or less independent projectors —the picture projector and the sound projector.

We will take up the picture projector first and then explain the necessity for the two separate projector systems.

Consider first the manner in which the film is exposed and then the manner in which it is projected. The negative enters and leaves the camera at a constant speed under the action of a uniform speed sprocket, but moves through the camera at an intermittent speed under the action of a claw-like arrangement (called the intermittent movement), controlled by a cam which engages the film and brings it to a stop in a position in front of the shutter. The shutter then opens and the film is exposed. The shutter then closes and during this closure the intermittent again engages the film moving that particular frame out of the aperture and bringing the next frame into position in front of the aperture. The variation in speed during the travel of the film through the camera is taken up in two loops formed by the film, one on each side of the intermittent movement.

This entire cycle takes place at high speed, there being 24 complete exposures per second in the camera.

After the film is processed, it must be projected in exactly the same way as exposed, the film entering the picture projector at a continuous speed, passing in front of the projector aperture under the action of an intermittent movement similar to the camera mechanism, and leaving the projector again at constant speed. For obvious reasons the sound track could not be projected in this manner, as one of the primary requisites for high-quality reproduction is a constantly uniform rate of speed of the film through the sound projector.

The sound projector is therefore placed behind (in point of time) the picture projector at sufficient distance to allow the film to be again given a constant rate of speed of movement, and explains the necessity for a separate projector for the sound. This distance is usually about

20 frames and is secured by synchronizing the sound this number of frames ahead of the picture when printing the composite positive from the picture and sound track negatives.

There have been several types of sound reproducing systems in use depending upon the type of record each is designed to reproduce. However, as sound on film is now used exclusively in motion pictures, and as the newest installations are capable of reproducing only the film record, the discussion in this text will be limited to this type.

Such a reproducing system consists essentially of a sound head, an amplifier system and a horn system.

The sound head, whether for the reproduction of variable density or variable area track, is made up of a light source, an optical system, and a photo-electric cell. The light rays from the light source act as a *conveyor* of energy while the sound track acts as a *control* of this energy and determines the amplitude and frequency of current change at the photo-electric cell. From this fluctuating current, an alternating voltage is derived, and after sufficient amplification, is delivered to the horns which reproduce the sound and project it into the auditorium.

1. LIGHT SOURCE AND OPTICAL SYSTEM

There are two general scanning methods, either direct or rear scanning, both consisting of a lamp whose light rays are carried through an optical system and then either through the film and the slit, in that or in the reverse order, and then through another set of lenses to the photo-electric cell.

Direct scanning is illustrated in Figure 87. The light passes through the optical unit and is focused on the scanning slit which projects a beam .084″ × .0013″ onto the film, and the part of the beam transmitted by the film is carried to the photocell by a second system of lenses.

Rear scanning is illustrated in Figure 89. Here the light beam, after passing through the condenser lens (whose focal point is some distance in front of the film), falls directly on the film and the blob of light transmitted is focused by the objective lens upon the scanning slit (.084″ x .0013″). The transmitted light then falls upon the photocell.

2. SLIT WIDTH

The width of the scanning slit has a great effect upon the quality of the reproduced sound, both from the reproduction as well as from the recording standpoint. If the slit is made too wide, the high-frequency response is lowered and distortion is introduced, while if too narrow,

the overall response is lowered and mechanical problems of manufacture are introduced.

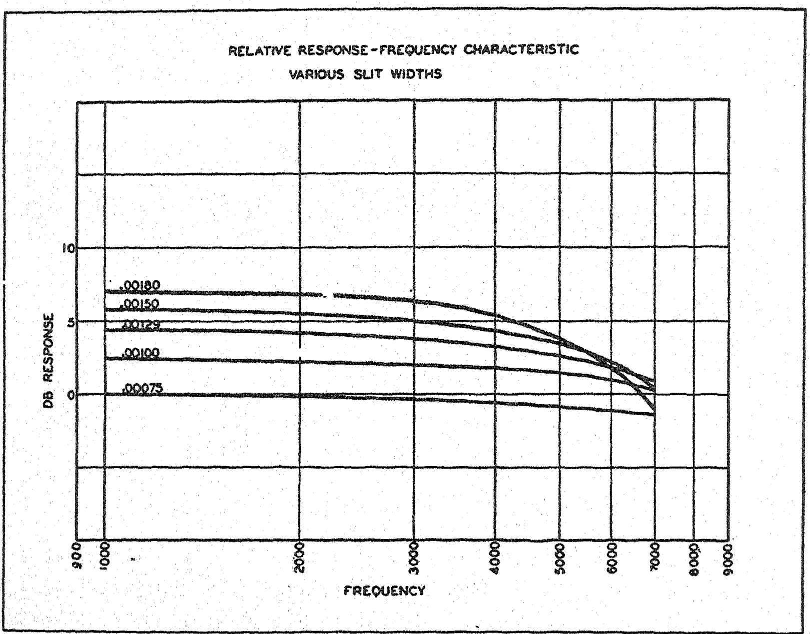


Figure 83.

Figure 83 shows the relative response from slits of different widths. The dimension of 0.0013″ was chosen as standard as giving the best compromise between the limiting factors mentioned above.

3. FILM VELOCITY

Investigation and experimentation have led to a great many improvements in the mechanical set-up of the sound reproducing system. Optical systems have been improved, vibration practically eliminated by better mechanical design, and the efficiency and response of the speaker system have been greatly improved.

Thus it was that, until very recently, one of the chief limitations to high-quality sound reproduction was the lack of constant velocity of the film as it moved past the scanning slit. Any variation in this speed of the film results in frequency modulation, which consists in a change in the relative frequencies of the signals on the film and leads to the reproduction of the signal itself as well as to the introduction of harmonics.

This is a form of distortion called flutter, and reduces the quality of the reproduced sound. However, the new-type sound heads now

available move the film at a very constant velocity and for all practical purposes, if properly adjusted, introduce no flutter.

4. PHOTO-ELECTRIC CELLS

Photo-electric cells fall under the general classification of photo-responsive devices, which are instruments responsive to radiant energy.

The term photo-electric cell, in its broadest sense, includes (1) photo-emissive cells, (2) photo-conducting cells, (3) photo-barrier cells, and (4) photo-voltaic cells. All these are similar in that they depend upon radiant energy for their action, but from each group a different reaction is secured.

In the first group, electrons are emitted from the cathode under the action of radiant energy, and collected upon the anode; in the second group, the resistance of the cell varies with the illumination; in the third group, the cell generates en e.m.f. under the action of the radiant energy; while in the fourth group, an e.m.f. is produced on one of two electrodes of certain materials when immersed in dilute electrolytes with one electrode illuminated.

Only the first group is of interest to us in this text, as this type is the only one used in motion picture sound reproduction, being far more sensitive than any of the other types in certain narrow regions of the spectrum and can be readily coupled to an amplifier system. Consequently, when the term photo-electric cell is used hereafter, this type cell is the one referred to. Photo-electric cells are responsive to radiant energy lying in or near the region of visible light, and act as a transducer of the energy, changing the radiant energy into electrical energy.

(a) Principle of Operation of a Photo-Electric Cell

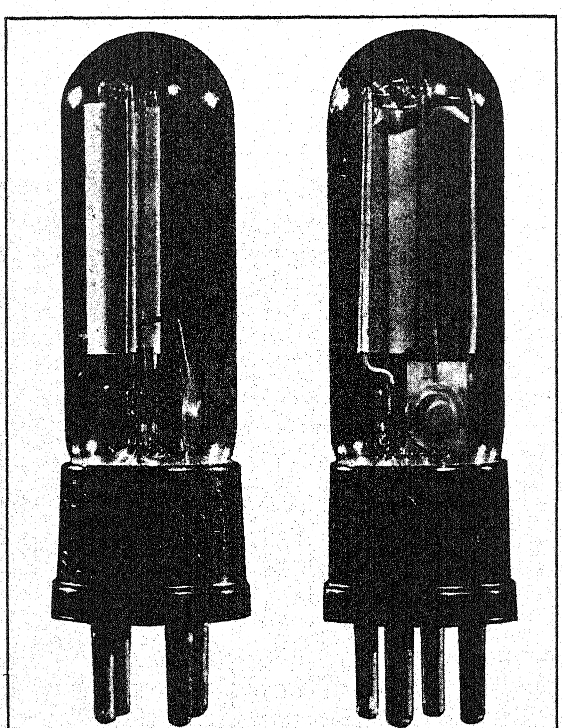
The principle of operation of photo-electric cells depends upon the fact that when radiant energy falls upon certain metals, electrons are caused to be emitted at a rate proportional to the total amount of illumination falling upon the surface. Consequently, if such a metal forms the cathode of an electric circuit, and another conductor is placed close to the metal, this second conductor will act as the anode of the circuit and a current will flow in the circuit. Thus, if properly applied, an instrument is available to change the "frozen" sound waves on the film to electrical waves, through the medium of light and the use of one of these sensitive metals.

The action of a photo-electric cell is similar to that of the vacuum tube in that electrons move through space from the cathode to the anode. However, in the vacuum tube, electrons are emitted from the cathode under the action of heat, while in a photo-electric cell they are emitted

under the action of radiant energy. The action of the vacuum tube is easily explained by the theories of physics and chemistry, but the action of the photo-electric cell is not so clearly depicted, as the same theories explain only in part the results obtained. Only at such time as the characteristics of light are more definitely known will the photo-electric phenomena be clearly understood.

(b) Photo-Electric Materials

Only a few metals emit electrons under the influence of light in any appreciable quantity. The most commonly used are the alkali metals: sodium, potassium, rubidium, and caesium, their sensitiveness decreasing in the order given. Also, each has a particular very narrow region in the spectrum where it is most sensitive, but the sensitiveness of any of these pure metals may be greatly increased by subjecting them to special treatments.



—*Courtesy RCA Manufacturing Co.*

Figure 84 — Two different views of a typical photo-electric cell used in sound reproduction.

There are two general types of photocells—those containing an inert gas and those which are evacuated.

In the latter the current consists totally of the electrons emitted from the cathode from the effect of the light falling upon the sensitive surface, but in the former the cathode-emitted electrons ionize the inert gas, and, as a consequence, when a sufficient voltage is applied the current is amplified within the cell itself.

The amount of this amplification is limited by the ionization voltage, that is, the voltage at which a visible glow discharge takes place, as this effect is self-perpetuating and injurious to the cell.

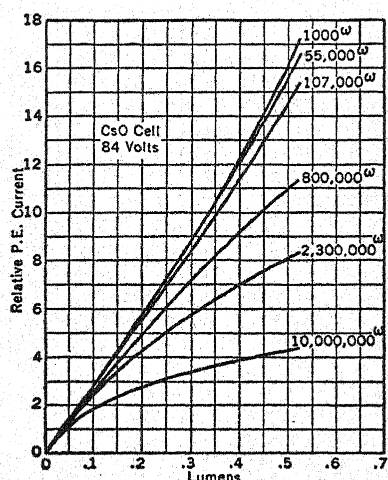
The effect of the illumination on the cell depends not only upon the material of which the cathode is made, but also upon the design of both the cathode and anode, as the photo-electric current depends upon the total illumination at any instant, and if the anode interferes with this illumination the current is reduced.

Two general types of construction are common: those with a central anode and those with a central cathode.

Figure 84 shows a central-anode cell, the type most generally used, with the cathode built in a semi-circular shape around the anode. The envelope is of glass or quartz, depending upon the metal used for the cathode and the source of illumination.

In an ideal cell, the photo-electric current is proportional to the total illumination but in practical applications of this principle the relationship is slightly non-linear because of the charging effect of the glass wall, interference of the anode, reflection effects, and other obscure phenomena not yet fully understood.

It is necessary that the impedance looking into the output of the cell be held to a low value to minimize this effect, but the impedance must also be high enough to insure protection against glow discharge. Figure 85 shows the relative current plotted against illumination at different values of terminating resistances.



—*Reprinted by permission from "Electrical Engineer's Handbook," by Pender & McIlwain, published by John Wiley & Sons, Inc.*

Figure 85 — Current-illumination relation for gas-filled photo-electric cell with various series resistances.

The photo-electric effect of any certain material depends upon the wave length of the light rays to which the cathode is exposed. In motion picture work a cathode of

caesium with a suitable light source is usually used. These cells are rated according to their output in amperes per lumen of steady incident light on the cathode with the type of light specified, or in terms of the slope of the anode current illumination curve.

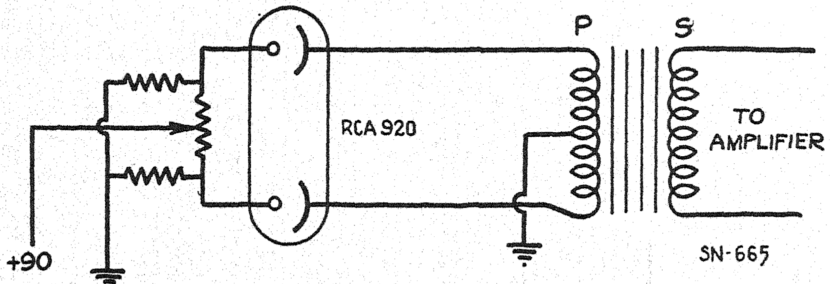
(c) Frequency Response

The frequency response of a commercial gas-filled photo-electric cell drops off in the higher-frequency range of the recording band, due to the presence of the gas in the cell and the capacitive effect between the electrodes and the circuit itself. This last effect becomes the limiting factor and determines the impedances which may be used.

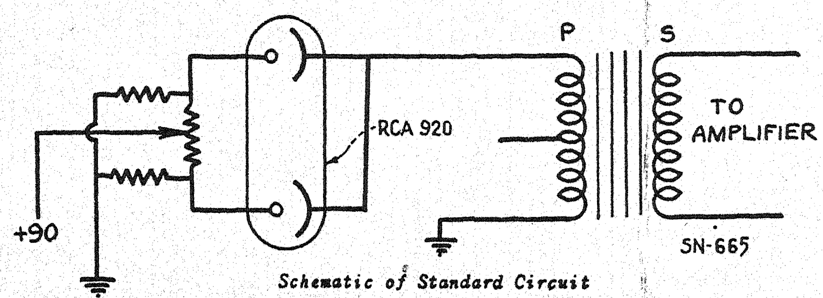
As the alternating-current output of the photo-electric cell is small, the circuit must be shielded or pick-up will result. The capacitive effect of the circuit necessitates close coupling with the first amplifier (P. E. C. amplifier).

5. PUSH-PULL SOUND HEADS WITH ASSOCIATED CIRCUITS

Figure 87 shows the schematic of an RCA MI-1070 sound head, known as a direct scanning reproducer, and consisting of a push-pull photocell, a special lens and prism assembly, together with a push-pull



Schematic of Push-Pull Circuit



Schematic of Standard Circuit

—*Courtesy RCA Manufacturing Co.*

Figure 86 — Schematics of RCA 920 Photocell for push-pull and single track reproduction.

photocell transformer. This combination permits the reproduction of either single track or push-pull recording.

The 920 photocell contains two anodes and two cathodes. When connected through the selector switch for single track reproduction, the cathodes are hooked up in parallel, and the photocell operates in the same manner as that of the standard cell. For push-pull reproduction, the two cathodes are separated and operate alternately through the photocell transformer.
(See Figure 87.)

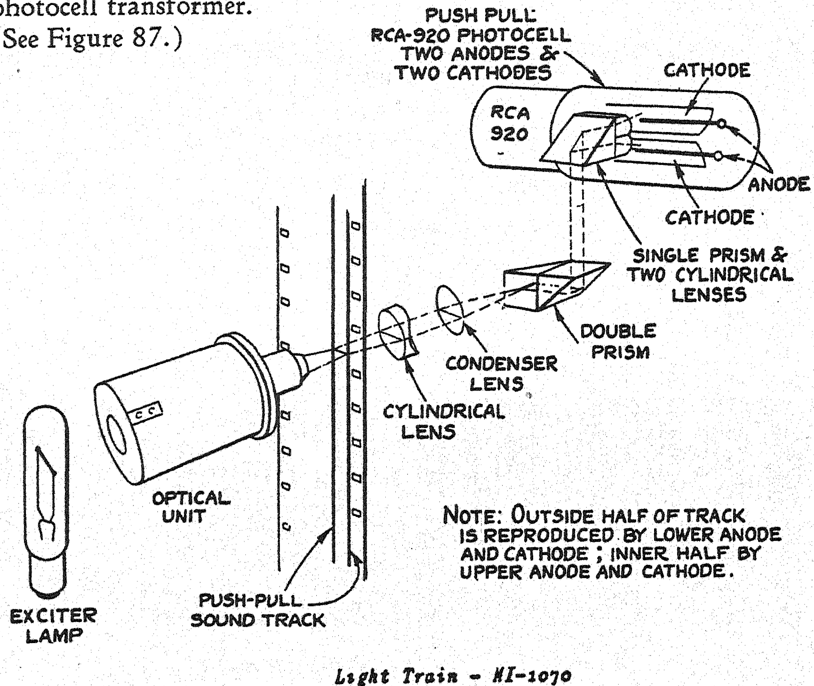


Figure 87 — Light train—MI-1070 sound head.

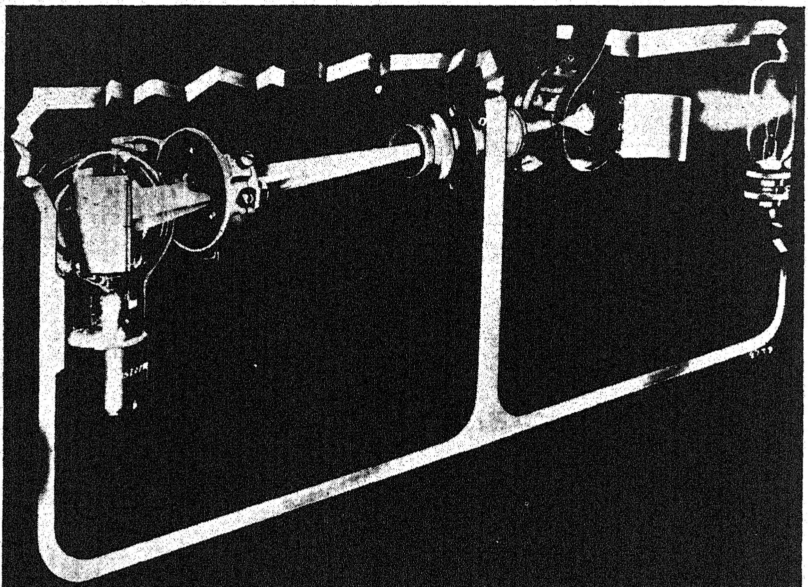
Any single track film may be used to balance this equipment for push-pull recording. When set for push-pull reproduction, the balancing potentiometer is operated to obtain minimum reproduction of the recorded sound at high volume.

Another method which is currently used in studio review room operation is to use alternating current as a source of supply for the exciter lamp and to adjust the balancing potentiometer for minimum output.

6. FLUTTER

To obtain the maximum benefit from increased volume range, flutter, due to the motion of the film at the point of scanning, must be reduced to an absolute minimum.

In the reproduction of sound recording, where a wheel with sprocket teeth is used to move the film, two types of flutter are encountered. The first is due to the irregular motion imparted to the film as it is engaged by the sprocket teeth and the second is due to the not-quite-constant speed of the sprocket wheel itself. For this reason scanners which use sprocket teeth have been eliminated in favor of drum-type equipment which practically eliminates the 96 cycle flutter. The drum also reduces low-frequency flutter when the rotary stabilizer principle is applied.*



—*Courtesy Electrical Research Products, Inc.*

Figure 88 — Western Electric sound head.

Figures 88 and 89 show another type of sound head manufactured by the Western Electric Company. This film scanning system is known as the "rear or indirect projection" type and consists essentially of an exciter lamp, condenser lens-prism assembly and objective lens, a scanning slit behind which is a collimating lens, and a photocell.

The condenser lens-prism assembly should be so adjusted that it focuses the filament image some distance in front of the film plane on the lamp side, so that the film is illuminated with a blob of light. The objective lens is adjusted to focus the track image sharply on the scanning slit, the width of which is approximately 1.3 mils.

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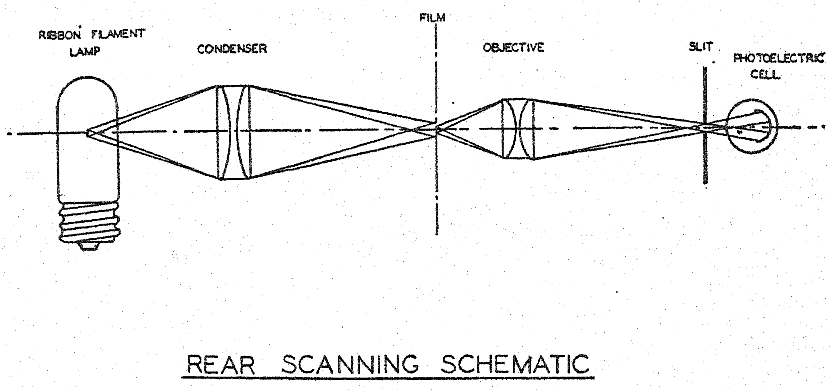
* Journal of Society of Motion Picture Engineers, October, 1935, "*Technical Aspects of the High-Fidelity Reproducer,*" E. D. Cook, pages 289-312.

The drum which holds the film in place at the point of scanning is of the rotary stabilizer type similar to that used in the RCA sound head.

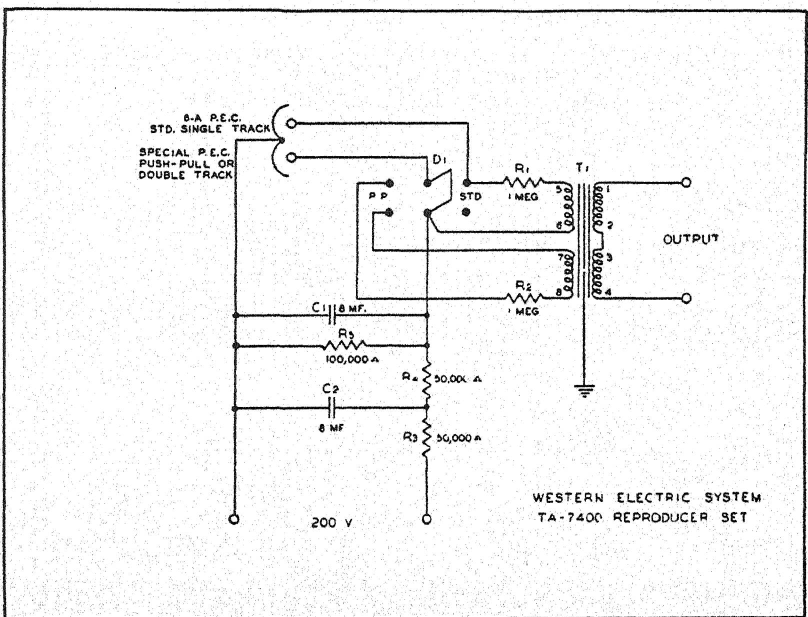
Figure 90 shows the wiring diagram of the TA-7400 reproducer set, for both push-pull and single track.



—*Courtesy Electrical Research Products, Inc.*

Figure 89 — Rear scanning schematic. Western Electric system.

The release of movietone push-pull sound track is very limited due to the small number of theatres equipped with push-pull reproducers. However, during the past year several pictures have been released with a



—*Courtesy Electrical Research Products, Inc.*

Figure 90 — Wiring diagram of TA-7400 reproducer set, Western Electric system.

limited number of push-pull copies. It is expected that within a short time enough theatres will be capable of playing push-pull so that it will be practicable to release push-pull prints on a larger scale.

7. OUTPUT POWER REQUIREMENTS FOR THEATRES

The use of "Hi-Range" release print sound tracks, now being released by some studios, requires that the theatre reproducer have suffi-

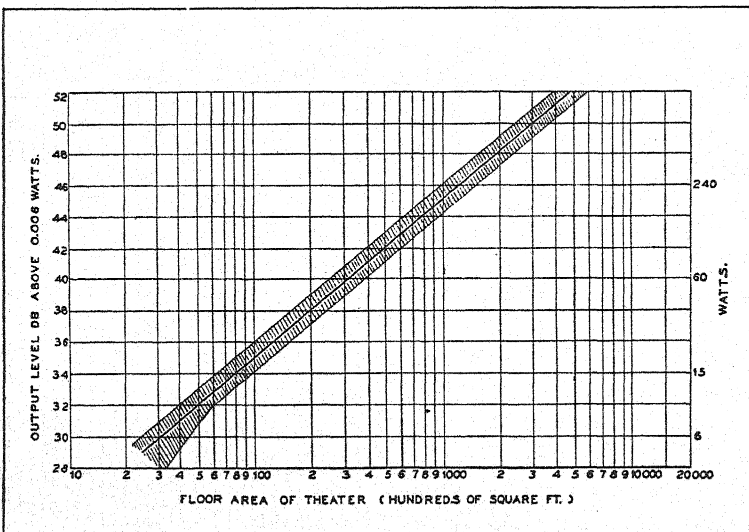


Figure 91 — Recommended amplifier output in electric watts in terms of theatre floor area in square feet.

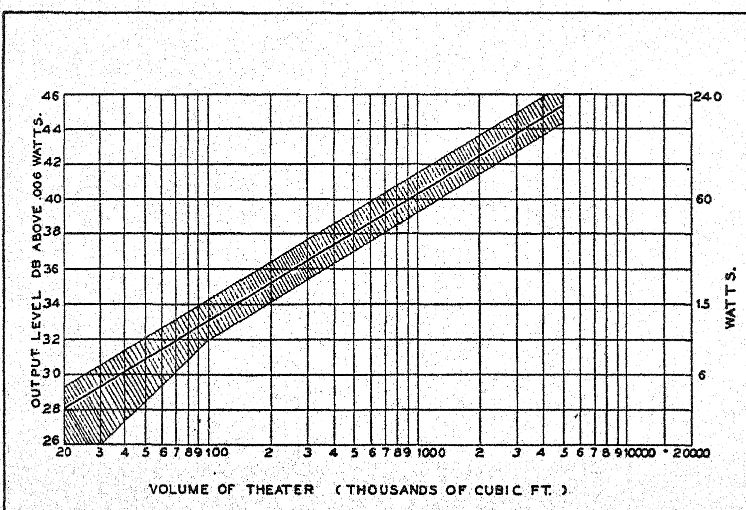


Figure 92 — Recommended amplifier output in electric watts in terms of the theatre volume in cubic feet.

cient output carrying capacity and efficiency to adequately reproduce this increased volume range without compression.

The history of the reproduction of sound has been one of continual increase in amplifier carrying capacity. Originally, output powers from two and one-half to twelve watts were considered adequate, but since then developments in recording have made it possible to utilize up to 60 db range, and it has consequently been found neces-

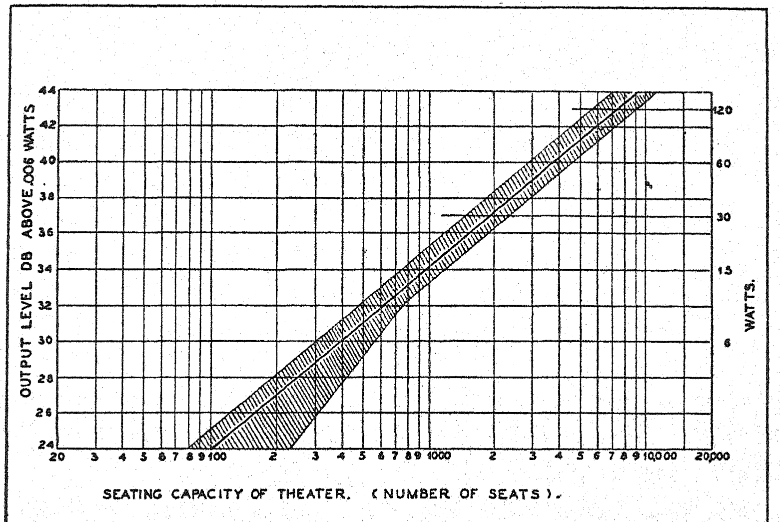
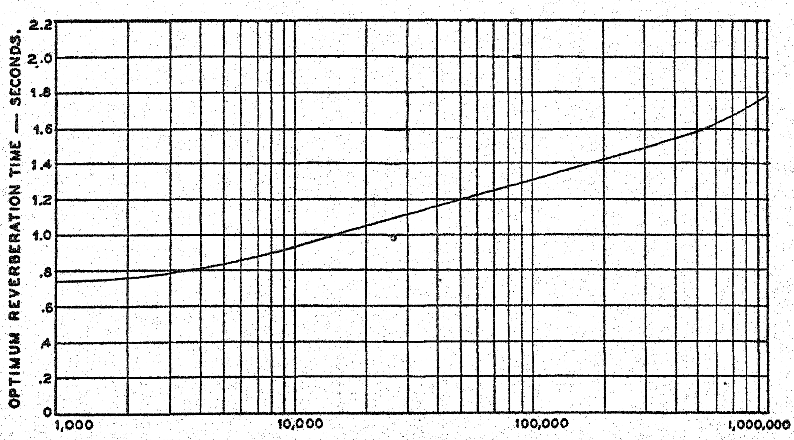


Figure 93 — Recommended amplifier output in electric watts in terms of the seating capacity of the theatre.



—*Courtesy Electrical Research Products, Inc.*

Figure 94 — Optimum reverberation times for different volumes in cubic feet for 512 cycles per second.

sary to increase the theatre equipment power-carrying capacity by large amounts.

Sound effects which incorporate screams, earthquakes, gunshots and other sound effects incident to warfare, demand sensation levels considerably higher than that which could be delivered in the past, and for this reason a maximum output level of not less than ninety sensation

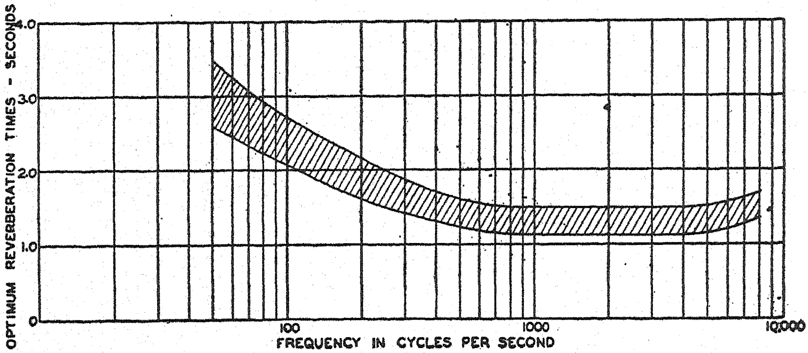


Figure 95 — Optimum reverberation times for different frequencies for a theatre of 300,000 cubic feet capacity.

units is now considered necessary, whereas in the past, amplifier carrying capacity has been limited to 80 db above the threshold of hearing.

8. VOLUME RANGE

High-quality theatre standards determine the necessity for a volume range of at least 60 db, which means that the residual noise of the

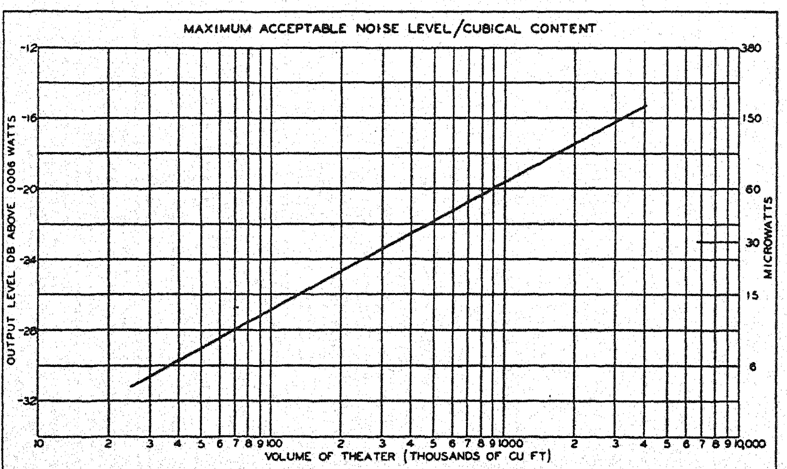


Figure 96 — Maximum acceptable noise level based on output level as recommended in Figure 92 (curve Figure 96, + 60 db = curve, Figure 92).

system, with no modulation, shall be not less than 60 db down from the maximum undistorted carrying capacity of the system. For example: A normal fifty watt system, which has a carrying capacity of plus forty db (6 milliwatt reference), should have a noise not greater than minus twenty, while smaller systems should have proportionately less noise.

Figure 96 shows the acceptable noise level for systems installed according to the recommendations given in Figure 92.

Figures 91, 92 and 93 indicate a yardstick to measure the amount of power necessary for a theatre, when either the floor space area, cubical content, or seating capacity is known. These curves, each of which is equally accurate, give the installed amplifier capacity necessary to maintain the standard required for high quality sound reproduction. Since the required power is a function of the absorption or reverberation in the theatre, deviation from these curves will be required, depending upon the permissible variation from optimum reverberation conditions.

The optimum reverberation time at 512 cycles per second is shown for auditoriums of various volumes in Figure 94. Figure 95 shows the optimum reverberation times in the recording frequency range for an auditorium of 300,000 cubic-feet volume.

One method by which this increased volume range may be procured is by the use of squeeze track on the release print sound track (see Chapter III).

Prints which have such a volume range are called "Hi-Range" prints, and the above mentioned increased amplifier power is necessary in the reproducing equipment to properly reproduce such prints.

Moreover, in reproducing a high volume print, the theatre manager and projectionist should follow the usual method of setting the fader for proper dialogue volume, which will automatically insure a proper reproduced volume level for any musical or sound effects passages in the same production. *If the volume level of the music or sound effects is reduced to a point lower than that originally intended at the time of the recording, dialogue passages will be too low for satisfactory reproduction.*

If the equipment is not functioning properly or if there is insufficient power capacity, the higher volume portions will reproduce with harshness and distortion.

The use of the higher amplifier power necessary to reproduce these prints also requires that the distribution of sound throughout the theatre be particularly uniform.

A "Hi-Range" print having a range of sound intensity of 50 db produces intensity changes which closely approximate those occurring in nature, and musical passages so recorded and subsequently reproduced with adequate power, lend the added color and naturalness necessary to insure a more complete enjoyment of the presentation.

9. STANDARD NOMENCLATURE FOR RELEASE PRINT SOUND TRACKS AND STANDARD FADER SETTING INSTRUCTION LEADER

As part of its program, the Research Council Committee on Standardization of Sound Projection Equipment Characteristics recently published papers on "Procedure for Projecting Hi-Range Prints," "Standard Nomenclature for Release Print Sound Tracks," and "Standard Fader Setting Instructions," which were distributed to every theatre in the United States, Canada and Alaska.

The Standard Nomenclature for Release Print Sound Tracks and the Standard Fader Setting Instructions as specified herein, have been formally approved by the Research Council of the Academy of Motion Picture Arts and Sciences, and adopted as standards for the motion picture industry, effective December 1, 1937.

Each of these papers is included in this book because of its timely interest.

STANDARD NOMENCLATURE for RELEASE PRINT SOUND TRACKS*

As a further step in the program of coordination between studio and theatre, the Research Council of the Academy of Motion Picture Arts and Sciences recently undertook to standardize the nomenclature for release print sound tracks, particularly as developments in sound recording equipment and technique have recently led to the appearance in the theatre field of a number of various new and different types of sound track.

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* Reprint from the Technical Bulletin of the Research Council of the Academy of Motion Picture Arts and Sciences, November 24, 1937.

SPECIFICATIONS

The Standard Nomenclature for Release Print Sound Tracks follows, with examples of each type included in the illustrations on the following pages.

Plays in "Std." Position of Sound Head Switch

Single variable density - - - - -	Figure 97
Single variable density squeeze - - -	Figure 98
Single variable density double squeeze - -	Figure 99
Unilateral variable area - - - - -	Figure 100
Bilateral variable area - - - - -	Figure 101
Duplex variable area - - - - -	Figure 102

Plays in "P.P." Position of Sound Head Switch

Push-pull variable density - - - -	Figure 103
Push-pull variable density squeeze - -	Figure 104
Push-pull variable area - - - - -	Figure 105

Classification as to Type of Recording

Figures 97, 98, 99, 103 and 104 on the following pages, illustrate the different types of variable density sound tracks, while Figures 100, 101, 102 and 105, illustrate the various variable area tracks.

As may be seen from the illustrations, these two general types of sound track differ fundamentally in that variable density recordings, either "single" or "push-pull," consist of alternate dark and light striations extending across the width of the track and gradually merging one into the other, the sound being represented by these changes in density, while the variable area recordings consist of black and clear transparent sections lengthwise of the film, the sound being represented by the wavy dividing line between these two sections.

Classification According to Power Requirements Necessary for Undistorted Reproduction

Those tracks illustrated in Figures 97, 100, 101, 102, 103 and 105, may be reproduced on those systems having a volume range which was considered adequate up to the present time and previous to the installation of the modern improved equipment with its relatively greater amplifier power.

Figure 97 — Single variable density.

Figure 98 — Single variable density squeeze, showing transition from full-width track to squeeze track.

Figure 99 — Single variable density double squeeze, showing transition from full-width track to double squeeze track.

Figure 100 — Unilateral variable area.

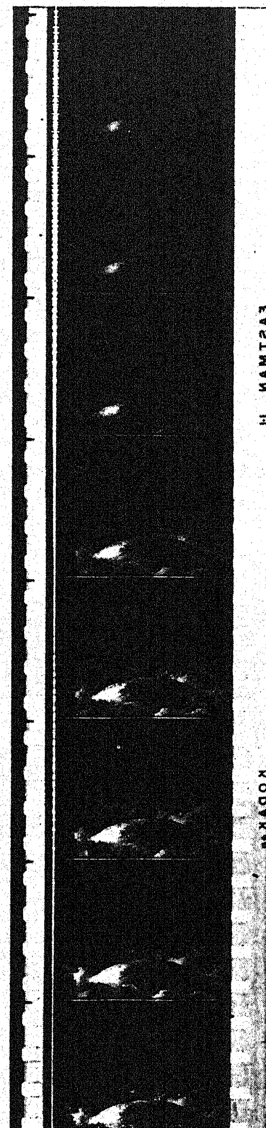

Figure 101 — Bilateral variable area.

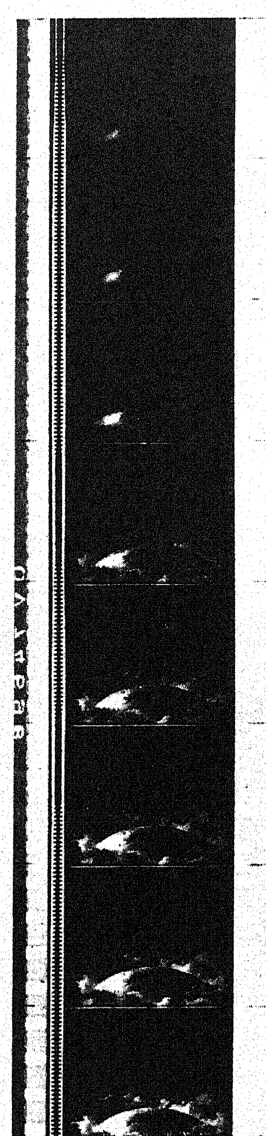

Figure 102 — Duplex variable area.

Figure 103 — Push-pull variable density.

Figure 104 — Push-pull variable density squeeze, showing full-width track, double squeeze of 6 db, and double squeeze of 12 db.

Figure 105 — Push-pull variable area.

Classification by Type of Equipment Necessary for Reproduction

"Push-pull" tracks as illustrated in Figures 103, 104 and 105, can be reproduced only on systems having a double or "push-pull" photocell, together with the necessary associated circuits.

Figure 104 illustrates the different amount of "squeeze," or track reduction, now being applied to variable density recordings. The upper portion of this figure shows a "push-pull" track before the application of any "squeeze," the center portion a reduction in track width of one-half, and the lower section a reduction of three-fourths, these being reductions of 6 and 12 db respectively.

STANDARD FADER SETTING INSTRUCTION LEADER*

To further aid the exhibitor in the proper handling of "Hi-Range" prints the studios, commencing about December 1, 1937, will utilize that part of the Academy Research Council Standard Release Print Leader which has been designated for use for any pertinent information to be transmitted from studio to theatre.

A portion of the specifications for the Standard Release Print Leader, indicating the location of this instructional information, is shown in Figure 106, with details of the information to be known as "Standard Fader Setting Instructions" being illustrated in Figure 107.

SPECIFICATIONS

The Standard Fader Setting Instruction Leader shall consist of 15 frames located as specified (Academy Research Council Standard Release Print Leader) in the synchronizing leader; the first frame shall designate the type of print; the second frame the type of reproducing equipment necessary to project the print; and the next nine frames the general fader setting specified in relation to an average fader setting for the particular product under consideration. The remaining frames may be used for whatever additional information the studio may wish to transmit to the theatre.

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* Reprinted from the Technical Bulletin of the Research Council of the Academy of Motion Picture Arts and Sciences, November 24, 1937.

This instruction leader will be of assistance to the exchange in that it will facilitate the special handling required in the exchange for the various types of prints, by providing an easily noted means of identification for each type.

It should be noted that the designation "Regular" in the Standard Fader Setting Instruction Leader indicates that only one type print has been issued on the particular production under consideration. Productions with prints designated as either "Hi-Range" or "Lo-Range" will have been issued in both type prints, i.e., all productions on "Hi-Range" prints will have necessarily been issued on "Lo-Range" prints as well.

This instruction leader will also enable the projectionist to identify a print which requires a "push-pull" reproducing system as contrasted to a print requiring a "single" system.

In order to identify more plainly the "push-pull" or "single" system prints, it was decided to include both the terms "push-pull" and "single" on every leader, crossing out in the laboratory one or the other of these two to leave the appropriate term designating the type sound track on the print. The illustration of the Standard Fader Setting Instruction Leader shown in Figure 107 indicates the manner by which this was accomplished for leaders which would be included in prints containing a sound track for reproduction on a "single" system. For leaders to be included in prints containing "push-pull" tracks the word "single" would have been crossed out, leaving the word "push-pull" to indicate this type of track.

In order that the exhibitor may achieve the best results, the fader setting designated in this leader should be followed in general, inasmuch as the entire balance between the dialogue and music throughout the reel will be chosen for each designated setting.

SPECIFICATIONS FOR

Academy Research Council Standard Release Print Leader Showing Location of Standard Fader Setting Instructions

Protective Leader

Shall be either transparent or raw stock.
When the protective leader has been reduced to a length of six feet it is to be restored to a length of eight feet.

Identification Leader (Part Title)

Shall contain 24 frames in each of which is plainly printed in black letters on white background: (a) type of print, (b) reel number (Arabic numeral not less than ¼ of frame height), and (c) picture title.

Synchronizing Leader

Shall consist of 20 frames ahead of Start mark, then 12 feet, including Start mark, to picture, opaque except as specified below: In the center of the first frame there shall be printed across the picture and sound track area a white line 1/32 inch wide upon which is superimposed a diamond ⅛ inch high.

The next 15 frames may be used by the studio for sensitometric or other information. If not so used this leader shall be opaque.

The Start mark shall be the 21st frame, in which is printed START (inverted) in black letters on white background. The Academy camera aperture height of .631 inch shall be used in the photography of this frame, and all others between Start mark and beginning of picture.

From the Start mark to the picture the leader shall contain frame lines which do not cross sound track area.

In the frames in which the numerals "6" and "9" appear, the words "six" and "nine" (also inverted) shall be placed immediately below the figure, to eliminate the possibility of mis-reading in the projection room due to the similarity between the inverted numerals.

Beginning 3 feet from the first frame of picture, each foot is to be plainly marked by a transparent frame containing an inverted black numeral at least ½ frame in height. Footage indicator numerals shall run consecutively from 3 to 11, inclusive. At a point exactly 20 frames ahead of the center of each footage numeral frame there shall be a diamond (white on black background) ⅛ inch high by ⅜ inch wide.

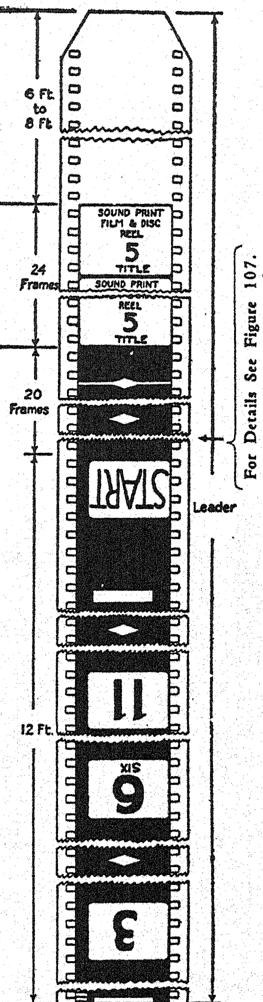


Figure 106.

Start of Picture

For specifications for motor and changeover cue location and reel end leader, see complete Academy Research Council Specifications for 35 mm. Motion Picture Release Prints in Standard 2000′ Lengths, published January 6, 1936.

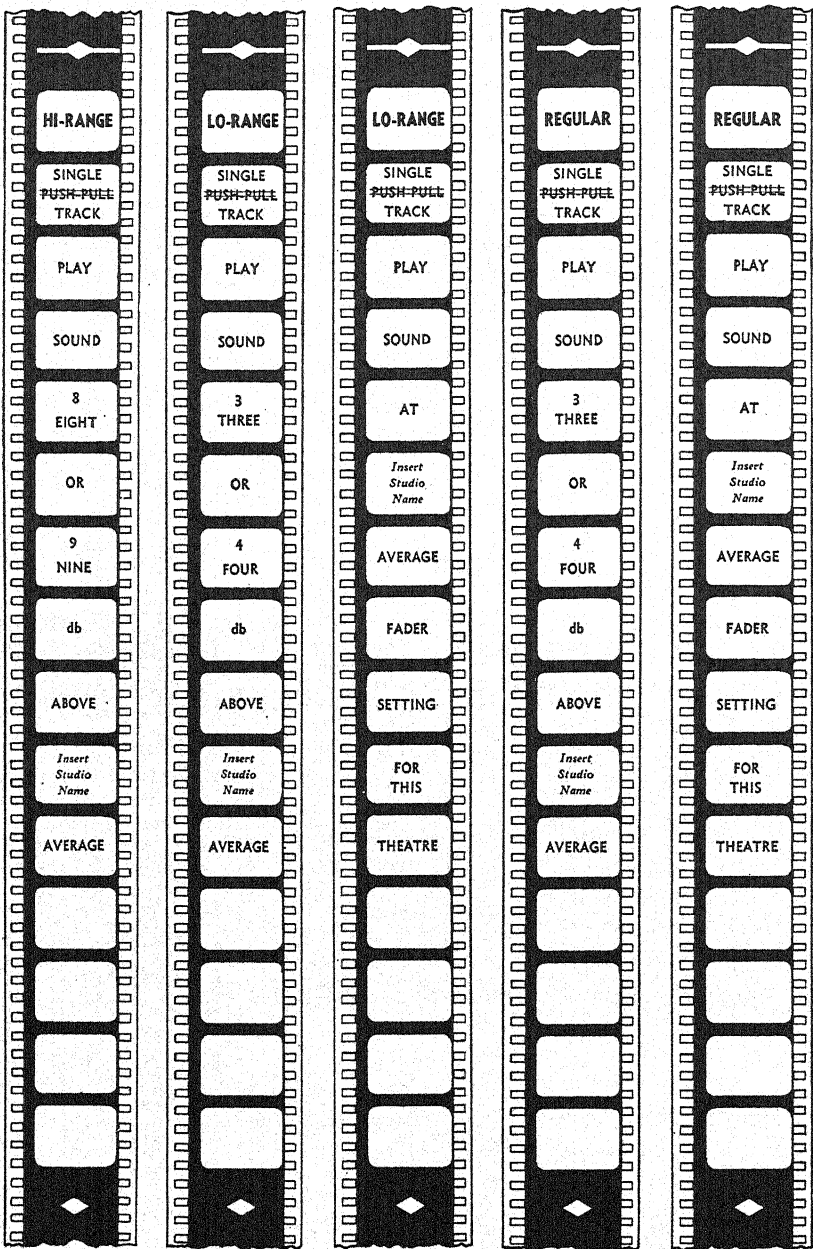


Figure 107 — Standard fader setting instruction leaders.

Chapter XI

SOUND CIRCUITS

By JOHN K. HILLIARD

1. THE REASON FOR CHOOSING PARTICULAR IMPEDANCE CIRCUITS

Many students of engineering often ask why, in communication circuits, 50, 200 or 500 ohm input and output impedances are used, and why in power circuits, a voltage of 110 is used. Inasmuch as the second question is more easily answered than the first, we will dispose of it by saying that there is no technical reason for a voltage of 110, other than that such voltage just happened to be selected, and has been used commonly for want of a good reason for changing it.

As for the first question, in voice and speech circuits, the length of circuit determines the loss-frequency characteristic for any given impedance. For very high impedances such as 10,000 ohms or more, the high frequencies (5,000 to 10,000 cycles) are attenuated much more rapidly than the low frequencies, due to the fact that the reactance of the capacity of the wiring is comparable to that of the circuit involved.

Consider the circuit illustrated in Figure 108. At the lower frequencies the reactance of C_G (which represents the stray capacities of the circuit and for purposes of illustration will be assumed to be 32 $\mu\mu$f) in shunt across R_L is so high that it has practically no effect on the power dissipated in R_L and can therefore be neglected. The maximum amplification of the circuit is then

Circuit 108 — Circuit illustrating the effect of capacitance at high frequencies.

$$\frac{E_L}{E_G} = \frac{R_L}{R_G + R_L} \tag{27}$$

At higher frequencies where the values of X_C become effective in the circuit, the extent of attenuation is indicated by the ratio of actual amplification at these higher frequencies to the maximum amplification.

$$\frac{\text{Actual amplification}}{\text{Maximum amplification}} = \frac{1}{\sqrt{1+\left(\dfrac{R}{X_C}\right)^2}} \qquad (28)^*$$

where $R =$ the equivalent resistance of R_G and R_L in parallel.

If now we assume that R_G and R_L are each equal to 1 megohm (10^6 ohms), then $R = \dfrac{R_G R_L}{R_G + R_L} = 500{,}000$ ohms and, at a frequency of 10,000 cycles, $X_C = 500{,}000$ ohms.

The ratio of actual to maximum amplification

$$= \frac{1}{\sqrt{1+\left(\dfrac{500{,}000}{500{,}000}\right)^2}} = \frac{1}{\sqrt{2}} = 0.707$$

or an attenuation of 3 db.

At 3,333 cycles the ratio is

$$\frac{1}{\sqrt{1+\left(\dfrac{500{,}000}{1{,}500{,}000}\right)^2}} = \frac{1}{\sqrt{\dfrac{10}{9}}} = \frac{3}{\sqrt{10}} = 0.95$$

or an attenuation of 0.5 db.

If R_G and R_L were taken as 500 ohms each, the ratio of actual to maximum amplification at 10,000 cycles

$$= \frac{1}{\sqrt{1+\left(\dfrac{1}{2000}\right)^2}}$$

The value of $\dfrac{R}{X_C}$ is thus so small as to be negligible and the ratio of actual to maximum amplification is practically one.

Thus we see a given capacity is less critical when circuit impedances are low. For this reason impedances of 500 ohms or less are practical values to use.

Where inductive pick-up is to be taken into consideration it is desirable to keep the impedance as high as possible or around 500 ohms, so that the voltage is high compared with the induced voltage. This is done in modern photocell output circuits by the use of a transformer, as shown in Figure 109.

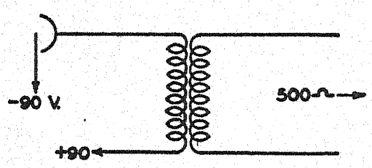


Figure 109 — Photocell output circuit coupled by a transformer.

* The derivation of equation (28) is shown on pages 178 and 179 of Terman's *"Radio Engineering,"* 2nd Edition.

2. CONDITIONS FOR MAXIMUM TRANSFER OF POWER FROM ONE CIRCUIT TO ANOTHER

The maximum power will be absorbed by one network from another which is coupled to it at two terminals, when the junction impedances looking into the two networks are conjugates of each other.

It can be demonstrated that the maximum power is absorbed from a generator when the external impedance is the conjugate of the internal impedance. Let us consider the case first where Z_G and Z_L are pure resistances. In Figure 111-A, where $Z_G = R_G$ and $Z_L = R_L$

$$I_L = \frac{E_G}{R_G + R_L}$$

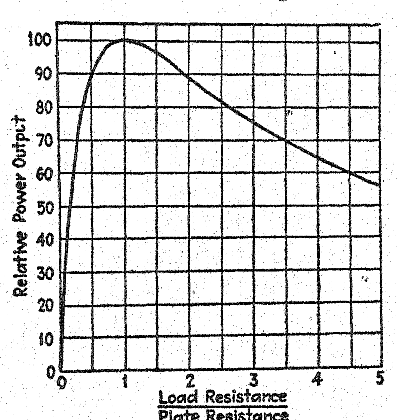
$$P_L = I_L^2 R_L = \frac{E_G^2 R_L}{(R_G + R_L)^2}$$

Differentiating the above equation and equating to zero to find the maximum power, gives

$$\frac{d P_L}{d R_L} = E_G^2 \frac{[(R_G + R_L)^2 - 2 R_L (R_G + R_L)]}{(R_G + R_L)^4} = 0$$

$$R_G^2 + 2R_G R_L + R_L^2 - 2R_G R_L - 2R_L^2 = 0$$

$$\therefore R_G = R_L \text{ (for maximum power transfer)}$$



—*From "Radio Engineering," by Frederick E. Terman (McGraw-Hill Book Co., Inc.).*

Figure 110 — Showing the relative power output at different ratios of generator and load resistances (maximum output when load resistance equals generator resistance).

If Z_G and Z_L have reactance then

$$P_L = \frac{E_G^2 R_L}{(R_G + R_L)^2 + (X_G + X_L)^2}$$

and it can be seen that as regards the reactance X, the power is a maximum when

$$X_L = -X_G \text{ (Resonance)}$$

which means that when Z_L is inductive, Z_G should be capacitive, indicating that a reactive load should be equal in impedance but negative in angle; that is, the impedances should be conjugates.

In power circuits, impedances are never matched to secure the greatest amount of power in the receiving circuit, since the efficiency under these conditions is only 50%. Power devices are rated on maximum possible load current within the safe heating range, and this safe current is considerably less than that which would flow if the circuit were matched. In recording circuits, however, where the cost of power is not a first consideration, and where currents are small so that temperature rise is negligible, considerable use is made of impedance matching to obtain maximum transfer of power from one circuit to another.

This brings us to what is known as the *reciprocity theorem,* which states that *when a source of voltage is connected across a pair of terminals of a passive four-terminal network, and an ammeter is connected across the other pair of terminals, the source of voltage and the ammeter may be interchanged without altering the reading of the ammeter.*

The effect of inserting a circuit element in a general network can be calculated by considering the current in the receiver before and after the element is inserted. The current in

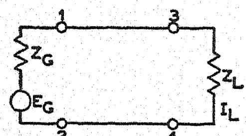


Figure 111-A

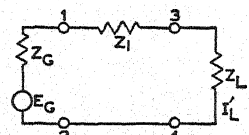


Figure 111-B

the receiver when the two circuits are connected together is

$$I_L = \frac{E_G}{Z_G + Z_L} \tag{29}$$

When a series impedance Z_1 is inserted between terminals 1-3, Figure 111-B, the receiver current is

$$I'_L = \frac{E_G}{Z_G + Z_L + Z_1} \tag{30}$$

Hence, the ratio of currents before and after the insertion of the series impedance is

$$\frac{I'_L}{I_L} = \frac{Z_G + Z_L}{Z_G + Z_L + Z_1} \tag{31}$$

Likewise, the current when Z_1 is connected in shunt across the terminals 1-2, Figure 112, is

$$I'_L = \frac{E_G\, Z_1}{Z_G\,(Z_L + Z_1) + Z_L\, Z_1}$$

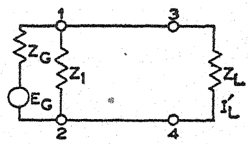


Figure 112.

$$\text{so } \frac{I'_L}{I_L} = \frac{\dfrac{Z_1}{Z_G\,(Z_L + Z_1) + Z_L\, Z_1}}{\dfrac{1}{Z_G + Z_L}}$$

$$\frac{I'_L}{I_L} = \frac{Z_1\,(Z_G + Z_L)}{Z_1\,(Z_G + Z_L) + Z_G\, Z_L} \tag{32}$$

NOTE—In the case of pure resistance, substitute R for Z in the above equations, and where Z is reactive it must be remembered that Z is a vector quantity.

Problems on the Effect of Inserting Impedances in a Network

Given: Circuit diagram Figure 111-A, where the generator and load impedances are resistances, i.e., $Z_G = R_G$ and $Z_L = R_L$.

$$R_G = R_L = 500 \text{ ohms}$$

1. What is the loss in db if a 500 ohm resistance R_1 is inserted in the circuit of Figure 111-A between terminals 1-3, as in Figure 111-B?

Let $I_L =$ the load current before insertion of the resistance.

$I'_L =$ the load current after insertion of the resistance.

$$\text{Then} \quad \frac{I'_L}{I_L} = \frac{R_G + R_L}{R_G + R_L + R_1} \tag{Eq. 31}$$

$$= \frac{1000}{1500} = 0.67$$

From Table VI, Page 455, an attenuation ratio of 0.67 indicates a loss of 3.5 db, or $N_{db} = 20 \log_{10} \dfrac{1}{0.67} = 3.5$ db.

2. What is the loss in db if R_1 of problem 1 is inserted in the circuit of Figure 111-A between terminals 1-2, as in Figure 112?

$$\frac{I'_L}{I_L} = \frac{R_1\,(R_G + R_L)}{R_1\,(R_G + R_L) + R_G\, R_L} \tag{Eq. 32}$$

$$= \frac{500\ (1000)}{500\ (1000) + 250{,}000} = \frac{500{,}000}{750{,}000} = 0.67$$

$$\text{Attenuation ratio} = 0.67$$

$$\text{Loss} = 3.5 \text{ db}$$

3. *Find:* The loss in db in the above circuit if an inductance of 1.6 henries is inserted between terminals 1-3 of Figure 111-A, at

(a) 50 cycles per second

(b) 10,000 cycles per second.

Given: Circuit diagram as in Figure 111-B where $Z_G = R_G$, $Z_L = R_L$ and $Z_1 = jX$.

$$R_G = R_L = 1{,}000 \text{ ohms; } L = 1.6 \text{ henries}$$

(a)

$$\frac{I'_L}{I_L} = \frac{R_G + R_L}{R_G + R_L + jX} \qquad \text{(Eq. 31, where } Z_1 = jX\text{)} \quad X = 2\pi fL$$

$$\frac{I'_L}{I_L} = \frac{1000 + 1000}{\sqrt{(1000 + 1000)^2 + (500)^2}} = \frac{2000}{\sqrt{4{,}250{,}000}}$$

$$= \frac{2000}{2064} = 0.969$$

so loss in db = 0.3

(b) at 10,000 cycles

$$X = 2\pi\, 1.6 \times 10{,}000 = 100{,}000$$

so $$\frac{I'_L}{I_L} = \frac{R_G + R_L}{R_G + R_L + jX} = \frac{2000}{\sqrt{(2000)^2 + (100{,}000)^2}}$$

$$= \frac{2000}{\sqrt{10{,}004{,}000{,}000}} = \frac{2000}{100{,}020}$$

$$= 0.02$$

so loss in db = 34 db

NOTE: This problem illustrates the relative loss due to an inductance in series with the load (at the approximate limits of the band used in recording) at low and high frequencies.

3. A STUDY OF POWER DELIVERED BY A CONSTANT VOLTAGE GENERATOR TO A LOAD CIRCUIT

If a battery has no internal resistance, R_G, that is, if $R_G = 0$, then I_{sc} = infinity, where I_{sc} = the short circuit current. Actually, however, this is never the case, for a battery always has some effective internal resistance. This resistance may be as low as 0.001 ohms for a huge storage battery, or up to one megohm or more for a silver chloride battery, depending upon the physical and chemical make-up of the battery.

We can, however, represent a generator or a source of power as a zero resistance generator with an accompanying external resistance, thus:

$$E = E_G - I_G R_G \tag{33}$$

Where E = effective voltage across the power source with circuit closed.

E_G = open circuit voltage across the power source.

I_G = the current flowing in the closed circuit.

This is known as the equivalent generator theorem and applies to any circuit from which power can be taken. (See Figure 113.)

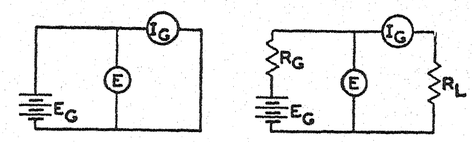


Figure 113 — Circuit diagram showing equivalent generator circuit.

If we let R_L = load resistance,

E_L = the voltage across the load,

I_L = the load current,

W_L = power consumed by the load,

$E_{oc} = E_G$

then it is found experimentally that as R_L is varied from infinity to zero, I_L varies from zero to I_{sc}, E_L varies from E_{oc} to zero and W_L ($= I_L E_L$) varies from zero to a maximum and back to zero. This maximum W_L occurs when $R_L = R_G$, at which time $E_L = \frac{1}{2} E_{oc}$; $I_L = \frac{E_{oc}}{2 R_L}$; and $W_L = W_G = \frac{E_{oc}^2}{4 R_L}$.

Here it will be noted that the condition for maximum output means $W_G = W_L$ and therefore just as much power is dissipated in the generator as in the load.

By definition, the voltage regulation of a system is

$$\text{Voltage Regulation (in per cent)} = \frac{E_{oc} - E_L}{E_L} 100 \tag{34}$$

Where

$R_G = R_L$, the voltage regulation is

$$\text{V.R.} = \frac{2E_L - E_L}{E_L} 100 = 100 \text{ per cent}$$

as

$$E_{oc} = 2E_L$$

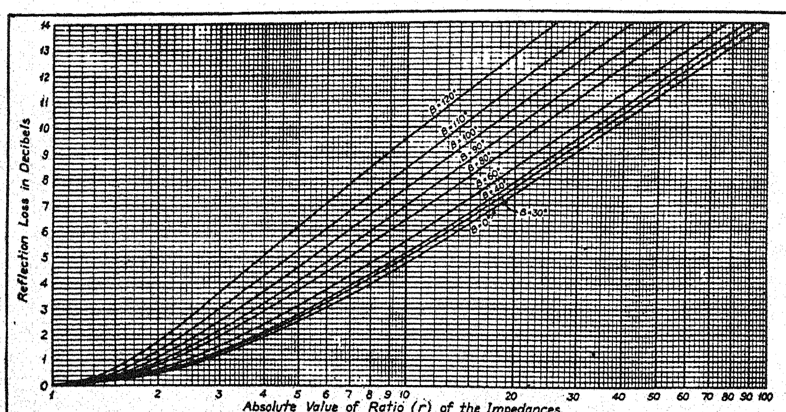
In the alternating-current case we must change our resistance to impedance, which means that we now have two independent variables instead of one, namely, the resistive and reactive components of the impedance.

In circuits which are not matched, reflection takes place at the point of mismatch. The amount of reflection will depend upon the ratio of the impedances looking in the two directions, that is

$$\text{Reflection factor} = \frac{\sqrt{4\, Z_G\, Z_L}}{Z_G + Z_L} \tag{35}$$

If $Z_G = Z_L$, the reflection factor $= 1$ (transmission factor).

The reflection in long circuits produces reversed currents, which may be out-of-phase sufficiently to nullify the incoming current, but in motion picture work this is of no appreciable value, since the time between the reflected wave and the original wave is in the order of a few microseconds, due to the short length of these circuits. However, when



The curves show the loss, in decibels, resulting from a mismatch of circuit impedances, as, for instance, connecting a 200-ohm line to a 500-ohm amplifier input.

The chart is a plot of the equation:

$$\text{Loss (in db)} = 10 \log_{10} \left[1 + \frac{(1 - r)^2}{4r \cos^2 \left(\frac{\beta}{2}\right)} \right]$$

Losses for values of β not given on the chart may be found by use of this equation.

Example of the use of the chart: Assume that a circuit having an impedance of $50\angle{-40°}$ is connected to a circuit with an impedance of $500\angle{130°}$. Adding the angles algebraically, we find $\beta = 90°$. The ratio r of the absolute values of the impedances is $500/50 = 10$. From the chart at the intersection of $\beta = 90°$ and $r = 10$ is read the loss 7 db, on the left ordinate.

Figure 114 — Losses in mismatched circuits.

working with vacuum tube output circuits feeding into a loud-speaker, the impedances are matched because certain types of vacuum tubes, such as pentodes, are critical to load impedance. A loud-speaker at its resonant point when working from a pentode, illustrates this type of matching, being able to absorb a great deal more power at its resonant point since its impedance is high.

A specific example of a case where the problem of mismatch between circuits is negligible, leading to no resultant distortion, is a mixer circuit, as it is only necessary that the output circuit from the amplifier be proper, and that the input to the following tube be such a value that the frequency characteristic is maintained.

The term, characteristic impedance, Z_o, is defined as

$$Z_o = \sqrt{Z_{oc} Z_{co}} \tag{36}$$

Hence, it is possible to find the impedance of a four-terminal network with an impedance bridge by measuring its open-circuit and closed-circuit impedances.

Example of Power Output Regulation Through Impedance Matching

1. Power Case:

If we assume that the open-circuit voltage across a generator is 115 volts, and that this voltage drops to 110 volts when a load drawing 30 amperes is connected to the generator, then:

$$Z_G = \frac{E_G - E_L}{I_L} = \frac{115 - 110}{30} = \frac{1}{6} = 0.167 \text{ ohm} \qquad \text{(Eq. 33)}$$

If now we were to match the generator and load impedances ($Z_L = Z_G$), then we would find that

$$I_L = \frac{E_G}{Z_G + Z_L} \qquad \text{(Eq. 29)}$$

$$\frac{115}{0.334} = 344 \text{ amperes}$$

which would burn out the equipment. It is therefore not always possible or advisable to match load and generator impedances in order to secure maximum power output.

2. Communication Case:

Assume $Z_G = 500$ ohms, $E_G = 20$ volts.

Then if

$Z_L = 10{,}000$ ohms	$I_L = 0.0019$ amp.	$E_L = 19.0$ volts	$P_L = 0.0361$ watts
$= 1{,}000$	$= 0.0133$	$= 13.3$	$= 0.177$
$= 500$	$= 0.020$	$= 10.0$	$= 0.20$
$= 200$	$= 0.0286$	$= 5.7$	$= 0.163$
$= 50$	$= 0.0364$	$= 1.8$	$= 0.0655$
$= 0$	$= 0.04$	$= 0$	$= 0$

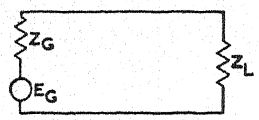


Figure 115 — Circuit diagram for Example 2.

FORMULAE

$$I_L = \frac{E_G}{Z_G + Z_L};\ E_G = Z_G\, I_L + Z_L\, I_L;$$

$$E_L = Z_L\, I_L;$$

$$E_L = \frac{Z_L}{Z_G + Z_L}\, E_G;\ P_L = E_L\, I_L$$

4. SHIELDING

Among the important and perplexing difficulties encountered in the design of recording and reproducing circuits are pick-up and cross-talk signals. Although the amount of energy encountered is very small it may prove extremely difficult to eliminate.

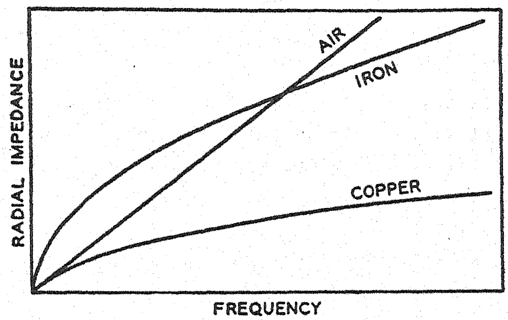
The problem of shielding is comparatively simple in principle but, in practice, radical steps are often necessary to eliminate pick-up and cross-talk.

One method of approach is based on the fact that the movement of energy at right angles to the circuits, which is the cause of pick-up, follows the same basic law as those laws which govern the desired transmission. From this point of view an electro-magnetic disturbance starts from the conductor and spreads radially outward throughout the surrounding space, constituting the first section of the radio transmission line. At the beginning and end of the shield the electrical characteristics of this line suddenly change. Outside of the shield the radio transmission line extends into space, but if the shield is correctly constructed very little energy is able to reach this section.

In their progress through metallic substances, electro-magnetic waves are attenuated at a rate depending upon the frequency, permeability, and conductivity of the metal. This attenuation is brought about by conversion of electrical energy into heat.

The effectiveness of a shield is due partly to the attenuation and partly to the reflection occurring at the boundary of the shield because of the mismatch, in radial impedance, of the shield and surrounding

insulator. As shown in Figure 116, the radial impedance of copper is very much lower than that of air. This means that while at any particular frequency there is no reflection between iron and air, there will be a



—*From the Bell System Technical Journal.*

Figure 116 — Curves showing the radial impedance of air, iron and copper at different frequencies.

large reflection between copper and air. For this reason, in thin shields the higher attenuation loss of iron may be more than offset by the greater reflection between copper and air. If a composite shield is made of alternate layers of copper and iron, the effectiveness of the shield is very much greater when the outside layers are made of copper, due to the fact that the reflection loss between copper and air is higher since the attenuation loss is independent of the arrangement of the layers.

Another interesting fact is that while non-magnetic shields become increasingly effective with increase of frequency, this is not always true with magnetic shields. At low frequencies magnetic shields are very efficient. As the frequency increases they sometimes become less effective but ultimately reach a minimum beyond which they improve again, so that at sufficiently high frequencies they are always better than non-magnetic shields. These characteristics are due to the manner in which the impedances mismatch, as previously explained.

In the case of non-magnetic shields the impedances of the shield and dielectric are always mismatched except for zero frequency by an amount which increases with the frequency as shown in Figure 116, while for magnetic shields the mismatch is large at low and high frequencies, but is small at certain intermediate points.*

5. LONGITUDINAL CURRENTS

Very often in recording or reproducing circuits, inductive interference is picked up where currents are induced into both sides of a

* Bell Laboratory Record, "*A Theory of Shielding*," by S. A. Schelkunoff, March, 1936, page 229-232.

circuit in the same direction in parallel paths, and the return path is by some other part of the circuit (such as through the ground). If the impedance of the return path to ground is high, cross-talk may enter the circuit. Cross-talk currents often have a considerable magnitude when the communication circuit is exposed to power lines, due to irregularities in balance in the former circuit.

The most effective way of reducing the longitudinal current is to provide a short-circuit path to ground for this current or to create an open-circuit to this longitudinal current without affecting the transverse circuit.

Accurately balanced transformers may be inserted at both ends of a circuit, with a ground at the mid-point of each transformer as indicated in Figure 117.

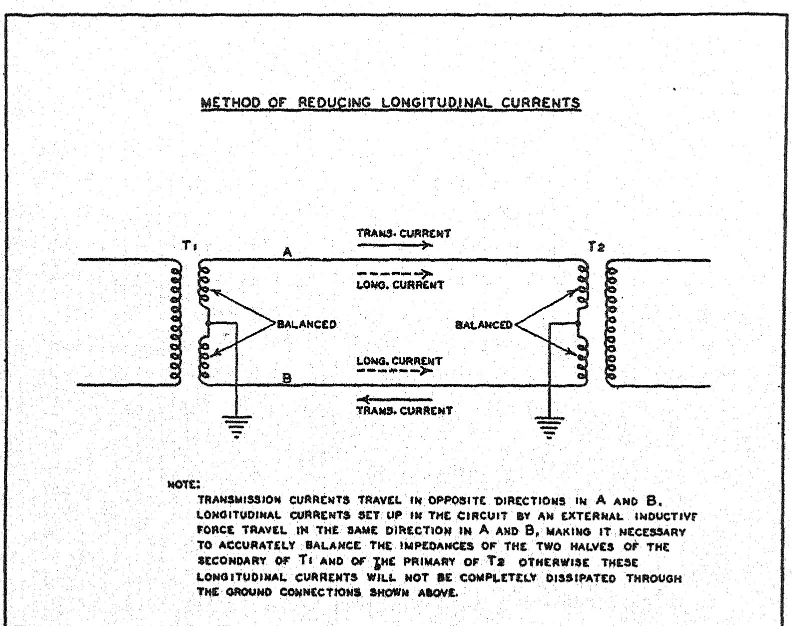


Figure 117 — Method of reducing longitudinal currents.

6. METHOD OF ACOUSTICALLY ADDING REVERBERATION TO SOUND TRACK

It is often desirable to increase the reverberation in an original sound track through the medium of recording.

During the earlier years of studio sound recording experience, this was accomplished by using staggered or offset tracks which were lined up a few frames apart and then mixed together to secure the desired effect.

Attempts were also made to use loud-speaker systems in highly reverberant echo chambers, combining in varying degrees the pick-up from them with the original sound. This system has been used to effect a different amount of "presence" in playbacks, depending upon the picture cut, to gain the proper perspective. However, when a large percentage of the sound is taken from the echo chamber, the high degree of quality required is not maintained due to the deficiencies in the loud speakers available.

Since the adoption of the current two-way loud-speaker systems, it has been found practical to re-record a sound track acoustically from an echo chamber. The distortion apparent when this track is compared back to back* with the original is a minor order effect, often not detectable at all. This is a very practical method of adding reverberation in recording without loss of the frequency characteristic of the original recording.

The necessary set-up consists of splitting the mixer into two banks, one bank for the control of those tracks which are intended to have reverberation added, the output of this bank being divided into two paths with isolation amplifiers. One path is directed into the echo chamber, the pick-up from this chamber appearing on one position of the

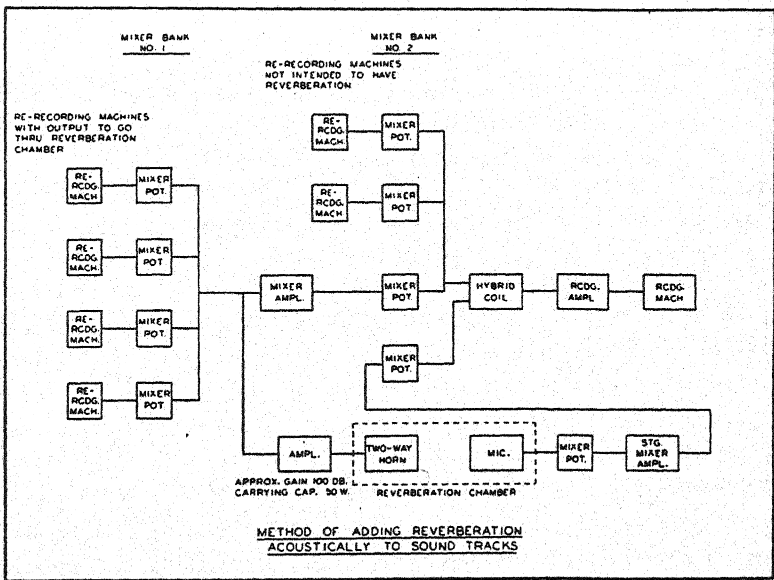


Figure 118.

*"Back to back" is a studio term for the direct comparison of two or more sound tracks run simultaneously on interlocked sound projection machines.

second mixer bank. The output of the second path of the first mixer bank appears also on the second bank. In this manner, the sound entering the echo chamber is pre-mixed and later combined with the original track to complete the desired illusion with the picture in terms of perspective. (See Figure 118.)

The echo chamber in which the horn and microphone are placed may be a room of approximately 10,000 cu. ft. volume, having a relatively high reverberation time. Studio experience indicates that a reverberation time of approximately five seconds is ample for any desired effect. This time period is obtained by lining the walls with a glazed hard surface material of considerable weight (sheet rock, or hard masonite covered with a hard paint has been found to be very practical), with the floor of painted wood or cement.

In order to obtain a flexible system of operation, so that more than one sound track may be passed through the chamber, the portion to have reverberation added must be "pre-mixed."

7. PRE-RECORDING

In order to maintain a degree of illusion of reality in music comparable to the degree of illusion of reality obtained in dialogue recording, somewhat more complex methods have been devised for recording music.

Practically all songs appearing in present day motion pictures are "pre-recorded," which means that the music is recorded before the actual filming of the picture. This is necessary in order that the continuity of the music may be maintained regardless of camera angle, which may shift in direct cuts from a long shot to medium shot or close-up and back, depending upon the dramatic requirements of the scene. Naturally, a steady flow of sound must accompany the picture regardless of the camera angle, and this steady continuation of music could not be maintained by direct recording at the time of the filming of the picture.

For "pre-recording," a play-back system has been devised, consisting of a portable disc or film reproducer, driven in step with the camera by means of either a synchronous or an interlock motor system, and used in conjunction with one or more horns located about the set.

The music previously recorded is played through this reproducing system and the artist, hearing the previously recorded song, provides the action by singing in front of the camera in tempo with the original recording. In this way a scene may be broken up into a series of short takes and need not necessarily be shot in its entirety at one time.

An automatic stop-start turn table, controlled by the director on the set, has proven a valuable tool in the use of play-backs.

The success of the pre-recording method depends upon the individual aptitude of the artist in providing well synchronized sound and picture, as well as adept editing of the sound track and picture to remove any out-of-sync condition.

A number of recordings of the same musical selection may be intercut to give the best possible rendition of the musical number. Instead of selecting a completed recording because it is the best overall average take, it is possible to intercut the best parts of a number of takes, thus reducing the effort on the part of the artist to record perfect complete takes, and considerably reducing the time required to obtain any given scene.

This technique of course requires a very high degree of precision of both artist and equipment, in order that exact pitch may be maintained between takes of the same number.

Chapter XII

MEASUREMENTS IN SOUND CIRCUITS

By JOHN K. HILLIARD

1. METHOD OF MEASURING HARMONICS

THE WEIN BRIDGE METHOD OF MEASURING HARMONICS: This bridge causes a network of resistances and condensers to completely balance out the fundamental frequency and when harmonics are introduced by an overloaded amplifier or a clashing light valve it is possible not only to hear the harmonics as they are being generated, but also to measure their amplitude.

2. LOAD CARRYING CAPACITY OF AMPLIFIERS

When the input voltage to an amplifier is increased to a point causing overload, the ouput voltage does not maintain the same wave form, and the distortion that results produces harmonics of frequencies not present in the signal being amplified. Usually in high-grade amplifiers when the output is more than 10 db down from its rated output, the percentage of harmonics is very small. For some time past it has been customary to rate the load carrying capacity of an amplifier at a value which produced 5% combined harmonics. Recently, however, this rating has been superseded by 1% total harmonics since high-grade reproducing equipment requires a much higher standard than formerly used. It has been customary to rate the amplifier in terms of the r.m.s. amplitude of the harmonics expressed as a percentage of the fundamental frequency component. The tendency will be to rate the amplifier in terms of the number of decibels difference between fundamental and harmonics. When making these harmonic measurements, it will be found desirable to check new set-ups at very low frequencies as well as at 1,000 cycles, since in most cases the carrying capacity is considerably reduced at the low frequencies because of transformers which have insufficient core material. (See Figure 119.)

3. METHODS OF MEASURING IMPEDANCE

If a source of frequencies such as an oscillator as well as either a power level indicator or a peak voltmeter and a variable resistance are

available, the resistive impedances of equipments may be measured in the field by the approximate method following. In the first method

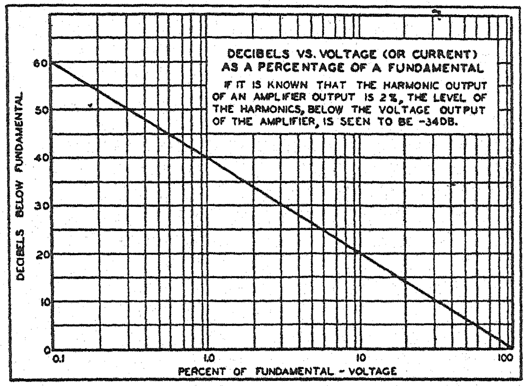


Figure 119 — Curve showing the per cent harmonics present in an amplifier.

(see Figure 120), which is of use in measuring load impedances, a calibrated variable resistance is placed between the generator and the load.

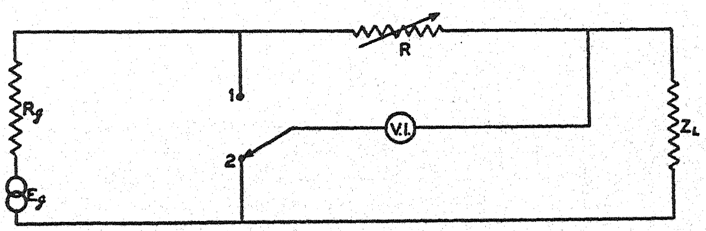


Figure 120 — Series method of measuring impedances.

The resistance is varied until, with the use of the peak voltmeter or power level indicator, the same voltage is obtained across the series re-

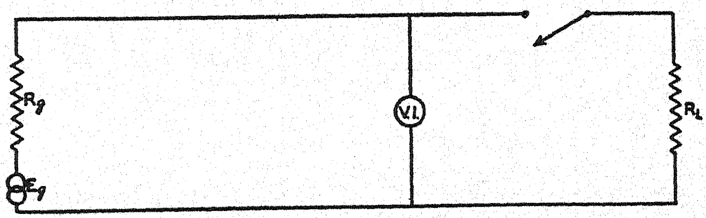


Figure 121 — Shunt method of measuring impedances.

sistance as is measured across the load. For this condition the load resistance R_L is qual to the amount indicated by the calibrated variable resistor.

The second method (see Figure 121) is useful in field work to measure the output impedance of amplifiers, oscillators, etc. It consists

of placing a voltage-indicating device across the generator terminals and measuring the output voltage with and without a known resistive load; that is, E_L and E_G respectively. For this condition $\frac{E_G}{E_L} = 1 + \frac{R_G}{R_L}$ from which R_G may be computed, since all other items of the equation are known. Where a volume indicator is used to indicate voltage, E_G and E_L may be obtained from the difference in db between the two level readings by means of the equation,

$$\text{db difference} = 20 \log \frac{E_G}{E_L}$$

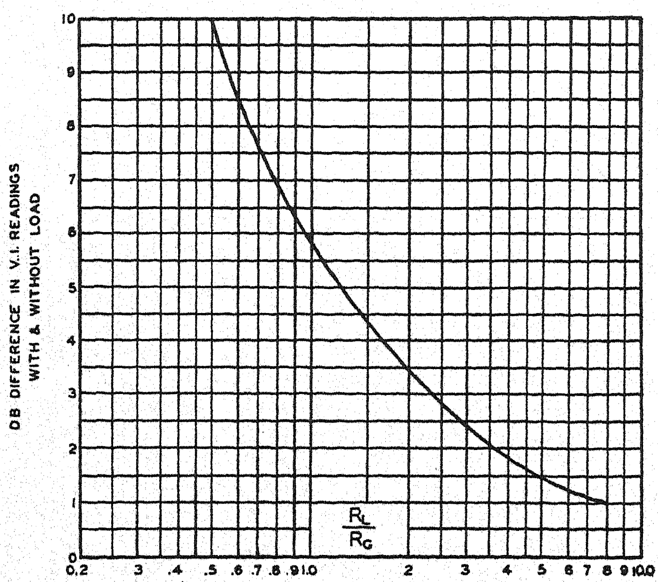


Figure 122 — Shunt method of measuring resistance impedance.

The curve of Figure 122 gives $\frac{R_L}{R_G}$ directly in terms of this difference reading in db. It is noted that where the known load impedance is equal to the generator impedance, the db difference reading is 6.0 db; that is

$$\frac{R_G}{R_L} = 1.$$

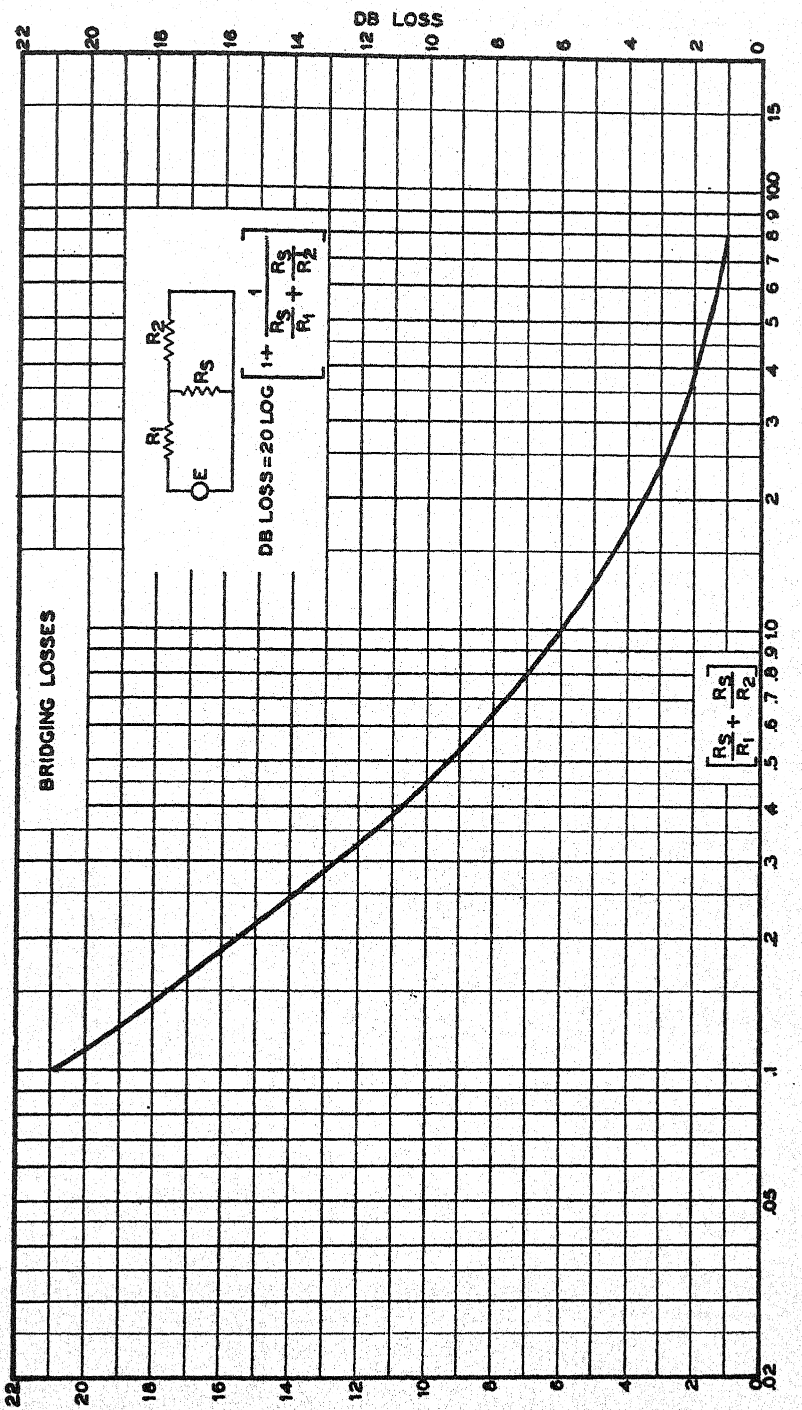

BRIDGING LOSSES
DB LOSS
DB LOSS = 20 LOG [1 + 1/(RS/R1 + RS/R2)]
[RS/R1 + RS/R2]
R1
R2
RS
E
22 20 18 16 14 12 10 8 6 4 2 0
.02 .05 .1 .2 .3 .4 .5 .6 .7 .8 .9 1.0 2 3 4 5 6 7 8 9 100 15

Figure 123.

GRAPHICAL SOLUTION OF BRIDGING LOSS

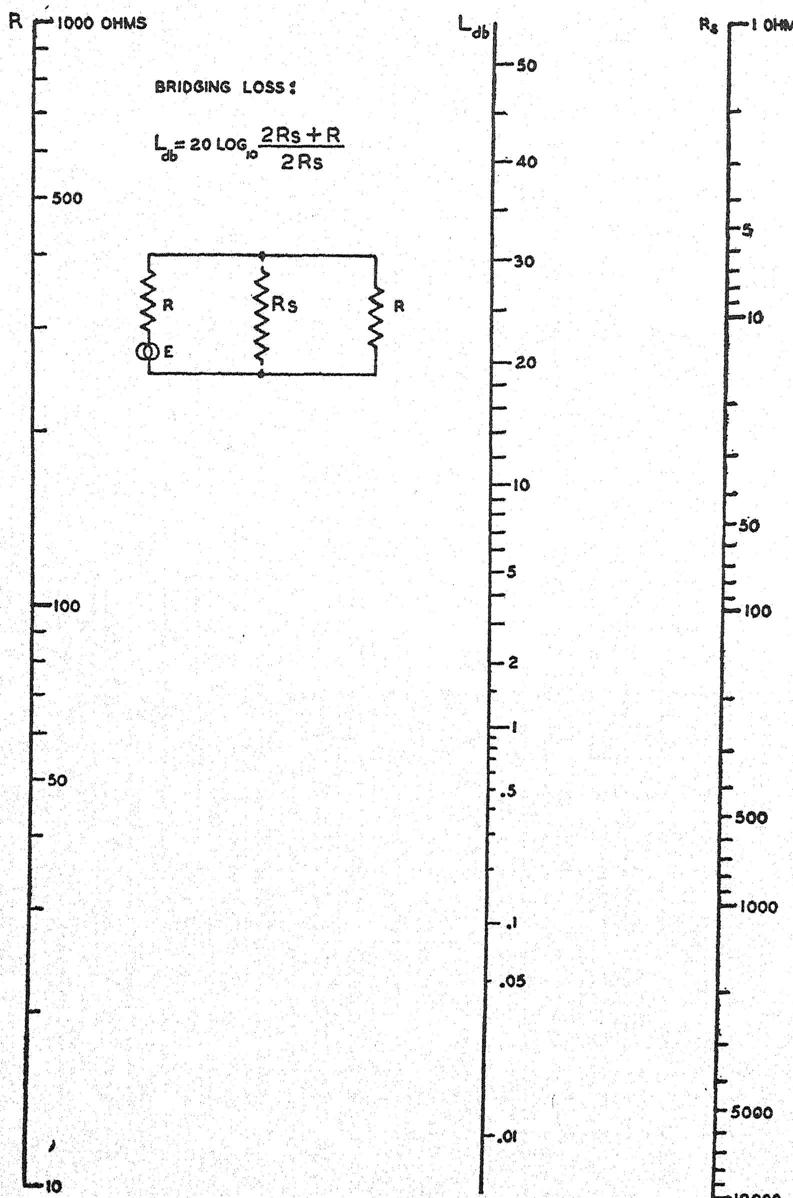


Figure 124 — Graphical solution of bridging loss.
(When R_1 and R_2 of Figure 123 are equal.)

When is it desired to measure the effect of bridging one resistive load across a circuit as shown in the bridging loss chart of Figure 123 or 124, the loss realized is equal to

$$20 \log \left[1 + \frac{1}{\frac{R_S}{R_1} + \frac{R_S}{R_2}} \right]$$

The curve of Figure 123 shows this loss plotted against the ratio

$$\left[\frac{R_S}{R_1} + \frac{R_S}{R_2} \right]$$

This curve may also be used for indicating the corrections for power level indicators when used on impedances other than those for which they are calibrated.

4. METHODS OF MEASURING DIVIDING NETWORKS

With the use of high quality, two-way loud speaker reproducing systems, which utilize dividing networks to distribute the power between the different bands, it is necessary to determine that the division of power is taking place properly. This is done by measuring their characteristics by the following method. The network is placed at the output of the amplifier from which it is to receive power, and the horn loads are removed and resistances equivalent to the speaker impedances substituted. A power level indicator is then placed across the low-frequency branch of the network, and a curve is obtained throughout the complete frequency range. The meter is then switched to the high-frequency leg and the procedure duplicated.

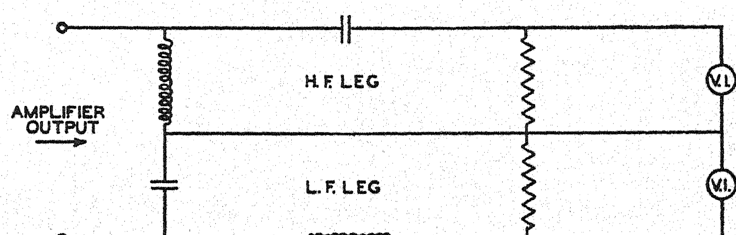


Figure 125 — Method of measuring dividing networks.

The usual half-section dividing networks now used in the field have an attenuation of 12 db per octave if the crossover point is at 300 cycles. This means that the attenuation in the low-frequency leg is 12 db down from 300 cycles at 600 cycles, and that the attenuation in the high-frequency leg is also down 12 db from 300 cycles at 150 cycles. In an efficient network of this type the minimum loss at the crossover point will be not less than 3 db and usually runs approximately 3½ db.

5. MICROPHONE MEASUREMENTS

(a) Distortion

The microphone introduces distortion into the system, depending upon the size of the microphone and the cavity resonance present in the microphone, the effect of the latter varying with the relation of the wave length of the sound to the dimensions of the cavity. Considerable research has been done in measuring these distortions, and it is a well known fact that the microphone and its enclosure should be made as small as is commercially practicable.

Due to the fact that the dynamic or moving coil type of microphone requires that the contents of the various elements of the circuit be so chosen that the magnitude of the impedances be the same at all frequencies, the microphone response is obtained by acoustic equalization. This equalization, which varies with frequency, usually takes the form of modifying the stiffness of the diaphragm, and providing vent or escapement tubes between the front and back of the diaphragm.

As a result of these various methods of equalization, it has proven difficult to manufacture these microphones commercially with a high degree of uniformity, while on the other hand, it is a comparatively easy task to so manufacture condenser microphones. Recently it has been found practicable to make small condenser microphones with a diameter of less than one inch and with gain-frequency and directional characteristics superior to those of the dynamic type. Since their size is small, distortion due to cavity and diaphragm resonance, and distortion due to field disturbance, has been greatly reduced.

In the near future it is very likely that this microphone will again come into popular use for dialogue recording.

(b) Microphone Calibration

In order to establish the characteristics of a microphone, it should be tested to determine its sensitivity, gain-frequency response characteristic, impedance-frequency characteristic, and directivity. Since sensitivity and gain-frequency characteristics can be obtained in terms of the pressure of the sound wave, the sensitivity is usually expressed in terms of the power output across a resistance load equal to that into which the microphone is designed to operate. The impedance characteristic will give a measure of the variations which arise when matching the microphone to the input of an amplifier.

A very practical method to obtain a pressure calibration of a condenser microphone, is to apply an electrostatic driving force to the

diaphragm, which can be done by mounting a fixed plate near the diaphragm, and applying an alternating voltage between the two surfaces.*

If two parallel plates are separated by a distance d cms, in air, and a statvoltage E is applied between them, the electrostatic pressure developed is

$$P = \frac{E^2}{8\,\pi\,d^2} \text{ dynes per sq. cm.} \qquad (37)$$

Now, if a sinusoidal voltage is applied where $E = e_m \sin \omega t.$, then

$$P = \frac{e_m{}^2 \sin^2 \omega t}{8\,\pi\,d^2} \text{ (instantaneous value of } P\text{)}$$

$$= e_m{}^2\,\frac{(1 - \cos 2\,\omega t)^2}{16\,\pi\,d^2}$$

which shows that the pure alternating-current voltage will develop a pressure of twice the frequency.

If the same alternating-current voltage ($e_m \sin \omega t$) were applied superimposed on a fixed direct-current voltage, e_0, the pressure developed will be

$$P = \frac{(e_0 + e_m \sin \omega t)^2}{8\,\pi\,d^2} \qquad (38)$$

as E is now equal to $(e_0 + e_m \sin \omega t)$.

This equation when reduced to practical values, that is, E in volts, will be

$$P = \frac{8.85\, e_0 \times e}{d^2}\, 10^{-7} \qquad (39)$$

where $P =$ r.m.s. pressure in dynes per sq. cm.
$e_0 =$ direct-current polarizing potential in volts
$e =$ r.m.s. value of the alternating-current component in volts.

In the case of ribbon, uni-directional and dynamic microphones, the field calibration is the most reliable type of measurement, and can be made by placing the microphone in the field of a loud-speaker whose characteristics are known, and then measuring the output of the microphone for all frequencies impressed on the loud-speaker. Since microphones and their associated amplifiers are subjected to considerable handling when used on a motion picture production, it has been found necessary to make routine measurements at frequent intervals.

A reliable check on such equipment consists of the following pro-

A source of voltage from the measuring set is introduced by means

of a transformer, or attenuator, or combination of both, with a very

* *Applied Acoustics*, Olson and Massa, pages 225-226.

cedure:

small resistance in series with the microphone. In this way, continuity is established between the microphone and its associated amplifier which will indicate poor contact, high resistance leads or decreased gain in the microphone amplifier. In the case of the ribbon microphone, the inserted resistance should be not greater than 1/10 of an ohm, while in the dynamic type of microphone, which usually has internal impedance of approximately 30 ohms, as much as one or two ohms can be placed in series without change in characteristic.

In the making of gain-frequency responses of certain component parts of a system, it is necessary that the test frequency have a good wave form, because of the fact that when filters are connected in a circuit, their true loss will not be indicated if the test frequency contains harmonics. For example: If a high-pass filter (which normally has a sharp cut-off at 100 cycles) is being tested, and a 50 cycle frequency with harmonics in it which are not more than 10 db down at 100 cycles, is impressed on the circuit, the filter will never show more than 10 db discrimination between these two points. Also, if oscillators having a high harmonic content are used for testing light valves, it is possible to clash the valve well below the over-load point at its resonant frequency even though it is presumably being tested at a lower frequency.

6. INTERMODULATION

Since various parts of the recording circuit are not strictly linear, tests are necessary from time to time to determine the degree of non-linearity. These can be conveniently made by applying two frequencies, such as 1,000 and 1,100 cycles simultaneously. The sum and difference frequencies will then appear due to intermodulation, and a measure of this effect can be obtained by filtering out the impressed frequencies and measuring either the sum or difference frequency.

7. ATTENUATOR DESIGN TABLES

Attenuator design is not a particularly difficult subject, but involves extensive computation. In order to avoid this computation, a fairly complete table of resistive values for all of the attenuator values in use is presented.

A few words about the choice of an attenuator for a given purpose:

When working into a high impedance device such as the grid of a tube, it is usually permissible to use a potentiometer, which is the simplest form of an attenuator. Here $R_1 + R_2 = Z_1$ (see diagrams and notations below) and since this sum is constant, only one switch arm and set of contacts will be needed to make a variable attenuator.

When working between two equal impedances in a circuit, such that it is not necessary that the impedance looking back into the output

terminals of the attenuator be equal to the load impedance (probably the most common case), an *L* type attenuator may be used. Here the equation given above no longer holds, and if the attenuator is to be variable, a double set of contacts and two switch arms will be necessary. Finally, when the attenuator impedance must match the load impedance, as in mixers and certain types of amplifiers with critical input impedances, a *T* type attenuator must be used. *T* type attenuators may be either symmetrical or unsymmetrical, depending upon whether the generator and load impedances are the same. (Table I includes only symmetrical *T* type attenuators. Unsymmetrical types are discussed later.)

All of the attenuators in Table I are computed for 500 ohm circuits; attenuators for other impedances can be computed very easily from the values given by multiplying all of the resistances by the ratio of the value of the impedance under consideration to 500 ohms.

For example, suppose that a symmetrical attenuator is desired which will give a 10 db loss in a 200 ohm circuit. Two hundred divided by five hundred equals four-tenths. From the table it is found that a 10 db 500 ohm *T* attenuator has two series resistances of 260 ohms each and a shunt resistance of 253 ohms. Multiplying each of these values by 0.4, gives a new value of 104 ohms for the series arm and 141 ohms for the shunt arm.

No values have been given for *H* type attenuators since they are easily derived from the corresponding *T* attenuator. The procedure is to halve each series resistor and to place the other half in the opposite leg.

The formulae from which all of the resistances given may be computed are fairly simple, but this is not the case for attenuators to work between unequal impedances. Since we now have two variables instead of one (i.e., both attenuation and impedances ratio), any reasonably complete table would be bulky.

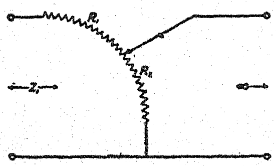
There is a minimum possible loss for which this type of attenuator can be constructed which depends upon impedance ratio. This loss varies from zero for a ratio of one, to infinity for a ratio of zero. See Chapter XXXIII for minimum loss attenuators of this type.

In conclusion, although all values have been given to three figures in the tables which follow, it is not necessary to hold to the actual values of the resistors given except in special cases where high accuracy is desired. A discrepancy of five per cent in any one resistance will cause an impedance mismatch of not more than that amount and a loss variation of only half a db.

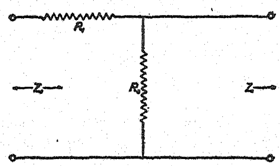
TABLE I

500 Ohm Attenuators

POTENTIOMETER

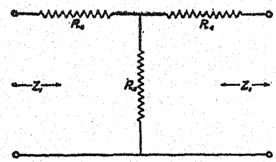


L TYPE

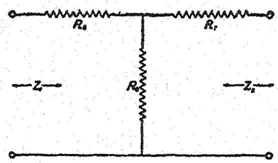


Attenuators

SYMMETRICAL



UNSYMMETRICAL



Loss, db.	R_1	R_2	R_3	R_4	R_5
1	54.5	445	4090	28.7	4340
2	103	397	1925	57.3	2160
3	146	354	1162	85.6	1420
4	185	315	855	113	1100
5	219	281	643	140	823
6	250	250	502	166	670
7	277	223	405	191	559
8	301	199	331	215	473
9	323	177	274	238	406
10	342	158	231	260	352
11	359	141	197	280	301
12	375	125	168	299	268
13	388	112	144	317	236
14	400	100	125	334	208
15	411	88.9	108	350	184
16	421	79.3	94.1	364	155
17	430	70.7	82.2	376	134
18	437	63.0	72.1	388	123
19	444	56.1	63.2	399	112
20	450	50.0	55.6	409	101
21	455	44.5	48.9	418	89.8
22	460	39.7	43.2	426	80.0
23	465	35.4	38.1	435	71.2
24	469	31.5	33.6	441	63.2
25	472	28.1	29.8	447	56.3
26	475	25.0	26.3	452	50.1
27	478	22.3	23.4	457	44.7
28	480	19.9	20.7	461	39.9
29	482	17.7	18.4	466	35.5
30	484	15.8	16.3	469	31.7
35	491	8.89	9.05	483	17.8
40	495	5.00	5.05	490	10.0
45	497	2.81	2.83	495	5.62
50	498	1.58	1.59	497	3.16
55	499	0.889	0.890	498	1.78
60	500	0.500	0.500	499	1.00

TABLE I (Continued)

Design Formulae

ATTENUATOR FORMULAE

POTENTIOMETER

$$R_1 = Z_1\ (1 - K)$$
$$R_2 = Z_1\ K$$

L ATTENUATOR

R_1 same as for potentiometer

$$R_3 = Z_1 \frac{K}{(1 - K)}$$
$$R_4 = Z_1 \frac{(1 - K)}{(1 + K)}$$

SYMMETRICAL *T* ATTENUATOR

$$R_5 = Z_1 \frac{2K}{(1 - K^2)}$$
$$R_6 = Z_1\ P_1 - P_2 \sqrt{Z_1\ Z_2}$$

UNSYMMETRICAL *T* ATTENUATOR

$$R_7 = Z_2\ P_1 - P_2 \sqrt{Z_1\ Z_2}$$
$$R_8 = P_2 \sqrt{Z_1\ Z_2}$$

The constants K, P_1 and P_2 in these formula are tabulated below. It should be noted that K is the voltage (or current) ratio corresponding to the given attenuation, and can be taken from any db-voltage ratio table or log table.

Loss, db	K	P_1	P_2
1	0.891	8.68	8.68
2	0.794	4.42	4.30
3	0.708	3.02	2.86
4	0.631	2.32	2.10
5	0.562	1.94	1.64
6	0.501	1.67	1.34
7	0.447	1.45	1.04
8	0.398	1.38	0.940
9	0.355	1.28	0.800
10	0.316	1.22	0.704
11	0.282	1.17	0.615
12	0.251	1.12	0.540
13	0.224	1.10	0.475
14	0.200	1.08	0.420
15	0.178	1.06	0.368
16	0.156	1.04	0.326
17	0.141	1.03	0.289
18	0.126	1.03	0.256
19	0.112	1.02	0.227
20	0.100	1.02	0.202
25	0.0562	1.00	0.112
30	0.0316	1.00	0.064
35	0.0178	1.00	0.036

—Data included in the above chart compiled by Dr. John F. Blackburn, Consulting Physicist, Hollywood, California.

Chapter XIII

PHASE DISTORTION

By JOHN K. HILLIARD

The composite waves of speech, music and sound effects are signals which have a wave composition quite different from those of single-frequency electrical waves. These complex compositions, which are ever changing in form at a very rapid rate (completely transient in character), are those parts of the wave train which have a bearing on the naturalness of reproduction. On the other hand, the waves show an indication of a definite steady state value after the transient part has subsided, and finally, again, we have a highly transient part of the wave beginning at the time the steady state is disturbed.

It is therefore necessary to retain all of the distinguishing qualities of a wave form to give natural reproduction of a sound. This can only be accomplished in an electro-acoustic system by making the frequency scale as wide as is commercially possible. This transient state transmission requires that a system accept steady state bands over an extremely wide frequency range.

In the reproduction of recorded material, certain distortion enters into the transmission to give imperfect results, which are due principally to three causes:

(1) The volume of the sound may be changed so as to be too loud or not loud enough.

(2) Amplitude distortion has taken place due to the fact that not all of the essential frequencies are transmitted with the same amplitude.

(3) Phase distortion is present to some extent in all recording systems. It is caused by the fact that the different frequencies composing the wave form travel with different velocities, such that their relative arrival times differ from their relative starting times. For undistorted transmission, the time of propagation of the various component frequencies must be the same.

To illustrate phase distortion; Figure 126 indicates the attenuation and delay characteristics of four band-pass filters in series. Note that the pass band is approximately 350 to 600 cycles. Now, in Figure 127, we see the oscillograms for several impulses of frequencies in the pass

and attenuation bands. Note that in each case there is a definite lag between the impulses sent and received, that the received signal in the pass band (at 480 cycles) is relatively undistorted, and that distortion

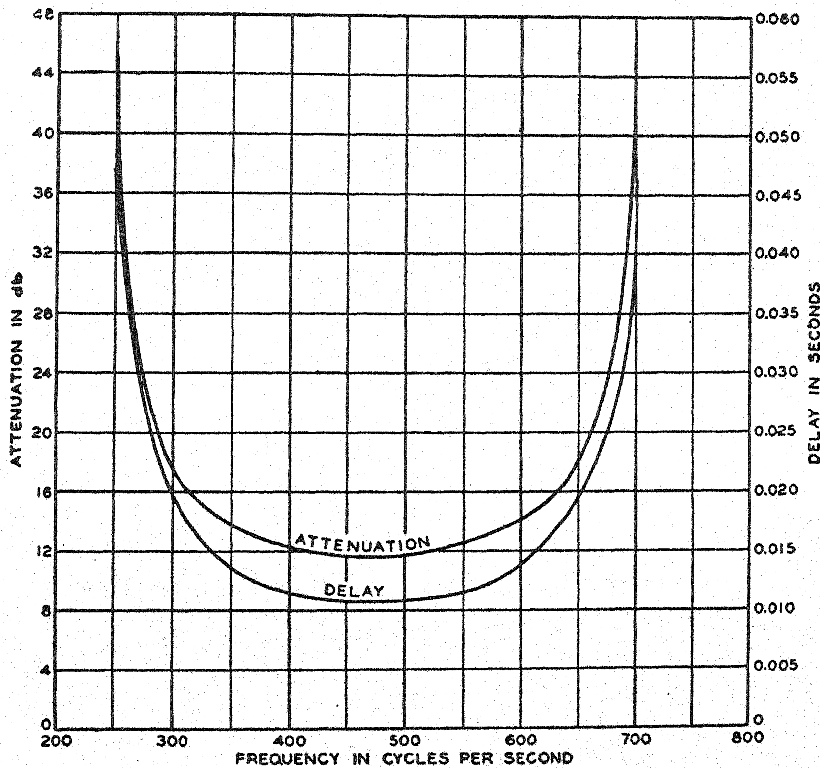


Figure 126 — Band-pass filter series — Attenuation and delay characteristics.

introduced near the edges of the pass band and in the attenuation regions is indicated by an increasing delay time in Figure 126. It will be observed that after the signal attains its steady state condition it is not distorted, and phase distortion is not present until this steady state is disturbed.

In general, we may say that the effect of phase distortion on speech is to decrease the intelligibility or to cause a loss of articulation. The effect on music is less noticeable due to the more sustained character of musical sound waves.

For transmission of transient waves without distortion, it has been shown that the time of propagation of component frequencies must be the same; this equality is obtained when the phase shift is linear with

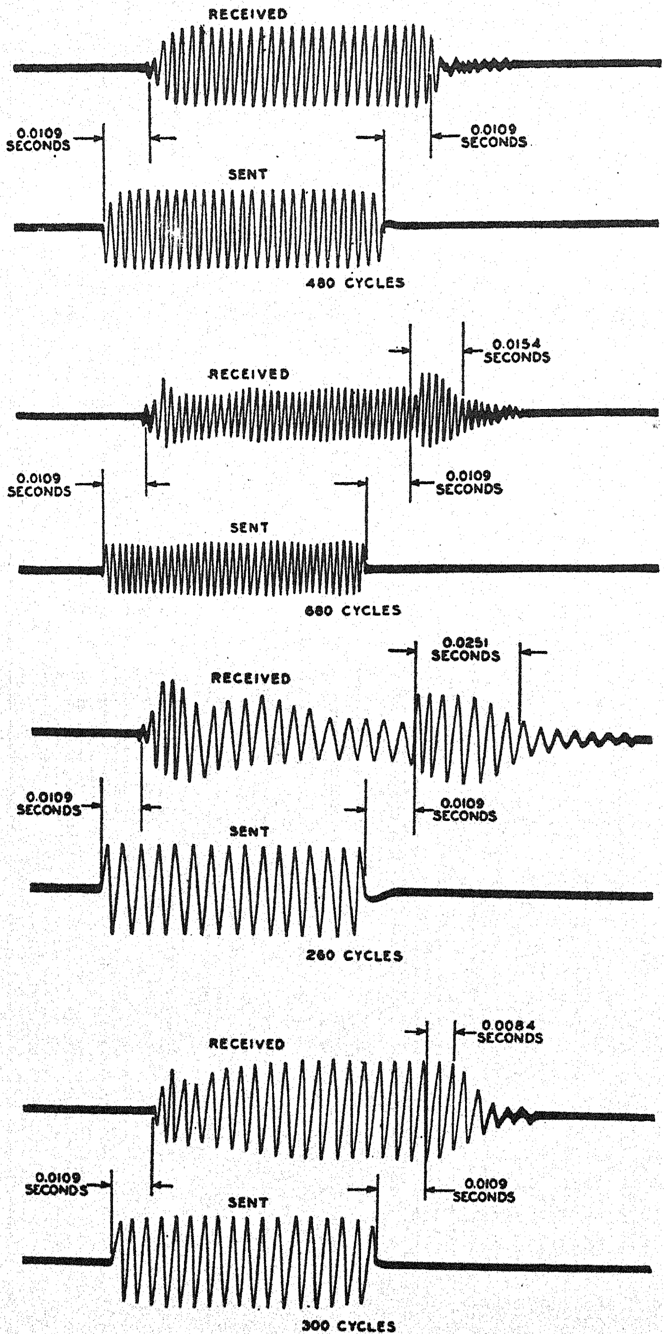


Figure 127 — Oscillograms — Frequencies of Figure 126.

frequency and equal to ($\pm n \pi$) when the frequency is zero, and n is equal to 0-1-2-3, etc. That is, when the phase shift is plotted as a function of frequency, the resulting graph should be a straight line. When a signal is applied to such a system, nothing will be received until a definite time period, after which period all waves will instantaneously assume their steady state values. The received signal will then be similar to the transmitted signal except for the *constant* phase shift introduced. This constant phase shift will not be detected by the ear. As phase changes in the component frequencies of steady state waves are not noticeable, we may assume that phase distortion occurs only in the transition periods.

Thus, a system may be linear in frequency response, as indicated by gain-frequency measurements (steady state), and yet introduce a distortion similar to non-linear systems under operating conditions.

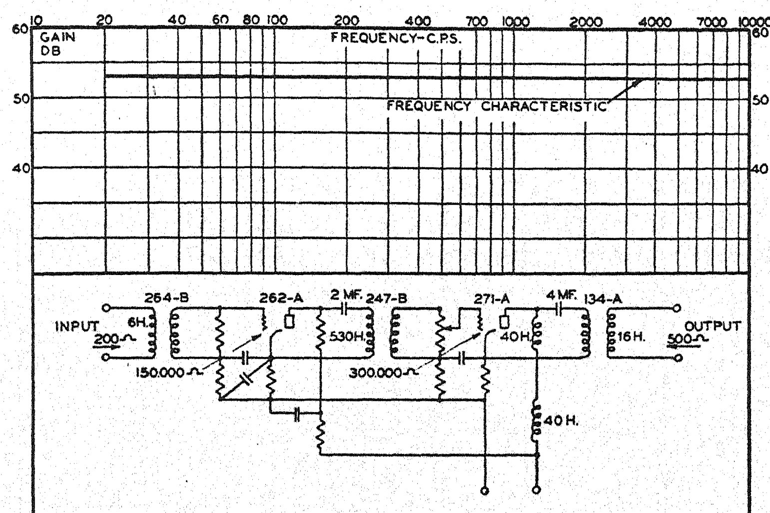
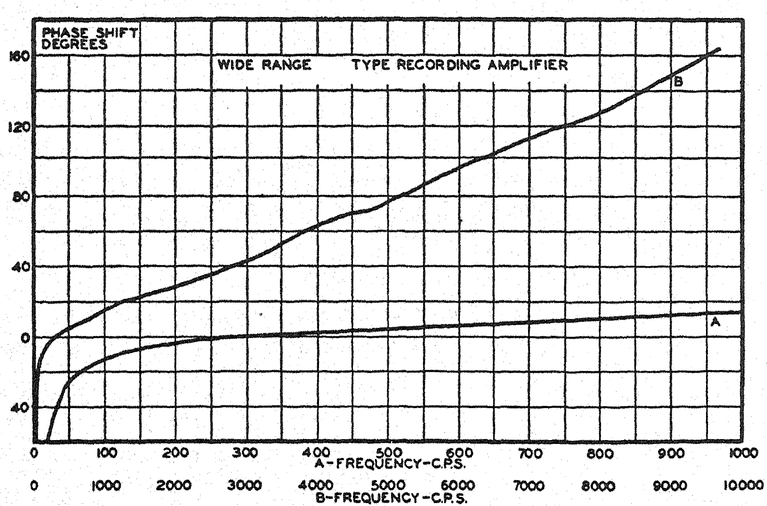


Figure 128 — Schematic diagram and frequency characteristic of a wide-range type recording amplifier.

Delay to high-frequency signals is more noticeable than to low frequencies due to the ear's dependence on the high frequencies for definition, etc. It has been determined that the maximum delay in the band from 5,000 to 8,000 c.p.s. should not exceed the 1,000 cycle delay time by more than 5 to 10 milliseconds. Also that at 50 c.p.s. the delay may be as much as 75 milliseconds more than the 1000 cycle value without noticeably affecting quality.

Both series capacity and shunt inductance delay low frequencies—the delay increasing as the values of such units are decreased. Conversely,

both series inductance and shunt capacity delay high frequencies—the delay increasing with inductance and capacity.



Upper scale for Curve A, lower scale for Curve B.

Figure 129 — Phase shift curve of Figure 128, 0 to 10,000 c.p.s.

However, we may say that in present high fidelity amplifiers, with their uniform frequency response, there is negligible delay distortion

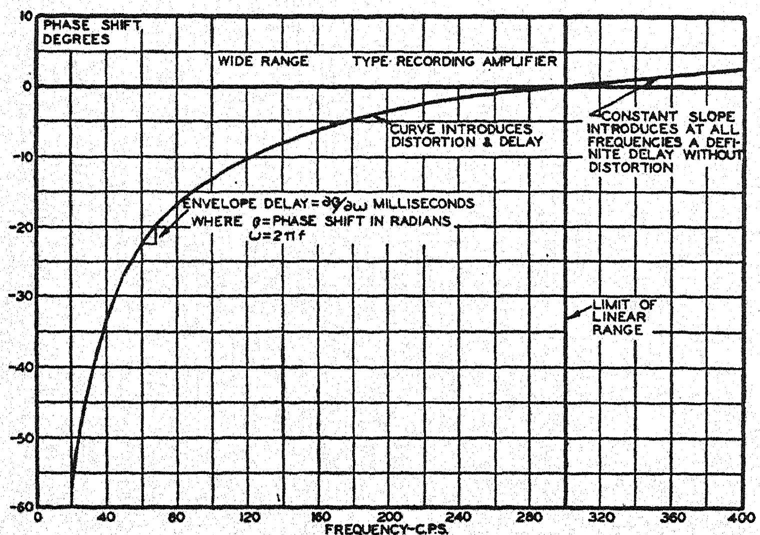
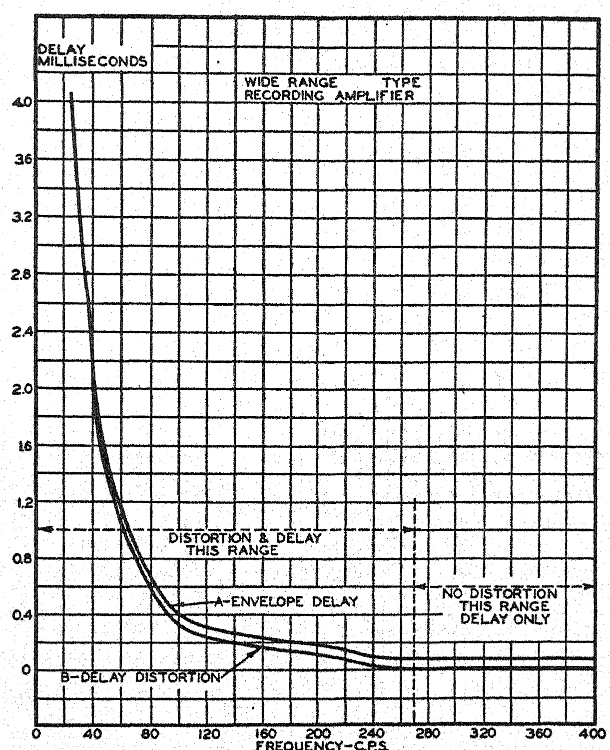


Figure 130 — Phase shift curve of Figure 128, 0 to 400 c.p.s.

in the relatively narrow frequency band employed. As an illustration we note in Figure 128 a wide range type recording amplifier with its fre-

quency characteristic. Observe the high values of shunt inductance and the large series condensers in this circuit. Now in Figure 129 the phase shift-frequency characteristic for this amplifier is seen to be relatively linear over the major portion of the frequency band. The slight irregularities or waves in the high-frequency portion are introduced by the measuring circuit, which will be described later. Observe that the devia-



Envelope delay in seconds $= d\,\beta / d\,\omega$

where $\beta =$ phase shift in radians

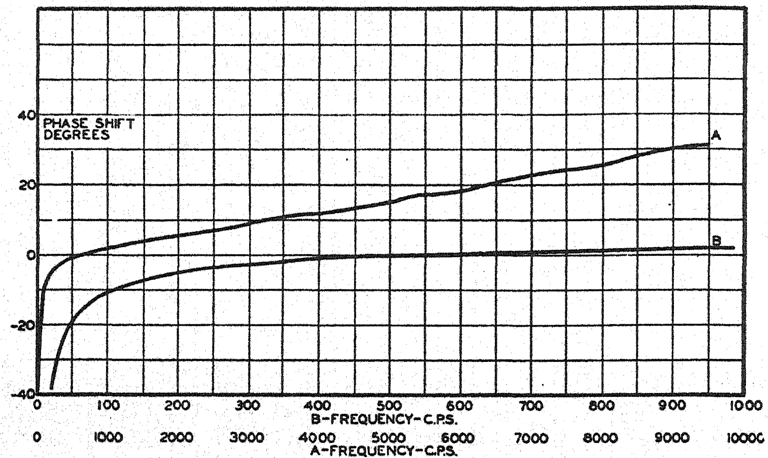
$\omega = 2\,\pi\,f$

Delay distortion at any frequency = envelope delay at that frequency minus envelope delay at frequencies where phase shift-frequency relation is linear.

Figure 131 — Delay distortion curve of Figure 128 from Figure 130.

tion from linearity is confined to the lower frequencies, which low-frequency section is shown in an enlarged scale in the lower curve. A further enlargement in Figure 130, which will be used later in the calculation of distortion delay, indicates that for all practical purposes we may consider the phase shift to be linear with frequency at all frequencies above 300 c.p.s. In Figure 131 we see the actual delay distortion in this range, below 300 c.p.s. Note that the delay distortion at 50 c.p.s.

is 1.4 milliseconds, and as we have previously stated that a delay distortion of 75 milliseconds at 50 c.p.s. is not noticeable, we see that the value of 1.4 milliseconds is negligible. This figure will be discussed in detail as we proceed to the actual measurement of delay distortion. Another phase shift curve is shown in Figure 132, which is the characteristic of a standard bridging amplifier. Note that the phase shift at 20 cycles with respect to 300 cycles is 35 degrees in contrast to 58 degrees for the wide-range type amplifier of Figure 128.



Upper scale for Curve B, lower scale for Curve A.

Figure 132 — Phase shift curve of bridging amplifier.

Proceeding to the theory of phase distortion we have the following proposition: "For an applied signal the *envelope of the oscillations* in response to an e.m.f. — $E \cos \omega t$, applied at time $t = 0$, reaches 50% of its ultimate steady state value at time, $t = \frac{d\beta}{d\omega}$, and its rate of building up is inversely proportional to $\sqrt{\frac{d^2\beta}{d\omega^2}}$. The quantity $\left(\frac{d\beta}{d\omega}\right)$ is defined as the envelope delay of a system in the range where the attenuation is not a function of frequency; that is, for a distortionless system with respect to frequency characteristic this quantity is the actual delay of the signal. Envelope delay is determined from the difference in the steady state phase shift for a definite interval of frequency, i.e., the received wave is delayed by a time interval given by the slope of the phase characteristic.

$$\text{Envelope delay} = t = \frac{d\beta}{d\omega} \qquad (40)$$

where $t =$ delay in seconds
$\beta =$ phase shift in radians
$\omega = 2\,\pi\,f$

Thus, referring again to Figure 130, we select a small change in frequency, f, and the corresponding change in phase shift is converted to β radians, and these two values are substituted in the formula. Solving the formula we obtain the time delay t for an intermediate frequency which we may assume to be $\left(\frac{f_1 + f_2}{2}\right)$. Proceeding in this manner we obtain the upper curve in Figure 131, which is the envelope delay. Now we note that above 300 c.p.s. the delay is constant and as it is not the actual value of delay, but the relative value from one frequency to another that is important, we subtract this constant delay from curve A, resulting in curve B, called the distortion delay curve. It should be noted that these results are in agreement with the statement that in order to have no phase distortion the phase shift must be linear with frequency (slope of phase shift curve constant) in order for the component frequencies of the signal to reach their approximate steady state condition in the same interval of time.

The major portion of phase shift in recording systems is due to the presence of filters. Delay distortion varies with the type of filter, number of sections and sharpness of cut-off; a sharp cut-off increasing the distortion in the pass band without greatly increasing the minimum delay.

We may briefly summarize the effects of different filter types as follows:

(1) In low-pass constant K filters, the phase shift increases with frequency from zero degrees at zero frequency to π radians at the cut-off frequency. The phase shift then remains constant above the cut-off frequency.

(2) In high-pass constant K filters the phase shift is constant at $(-\pi)$ radians for all frequencies below the cut-off and gradually passes from $(-\pi)$ radians to zero at infinite frequency.

In constant K type filters the phase shift is constant only in the attenuation bands and is changing with frequency over the entire transmission band. For full sections the phase shift in the attenuation band is always $(\pm\, n\,\pi)$ radians where n equals an integer. For half-sections the phase shift is one-half that of the full sections.

(3) In m-derived filters the phase shift slope in the pass band is usually less than for constant K types and the curvature is more pronounced near the edge of the attenuation band. The phase shift is not

always ($\pm \pi$) radians per section as for the constant K filter, but has this value only between the cut-off frequency and the frequency of infinite attenuation. The phase shift is zero for all other frequencies in the attenuation band.

(4) Phase shift characteristics of band-pass filters vary widely with the type of filter, but in general the amount of phase distortion is inversely proportional to the band width and is independent of the position of this band in the frequency scale. Conditions at the low-frequency end are similar to high-pass filters and at the high-frequency end similar to low-pass filters. If the straight portion of the band-pass filter phase shift curve is extended, it may intersect the phase shift axis at any point, where in low-pass filters the intersection is always at ($n\pi$) radians.

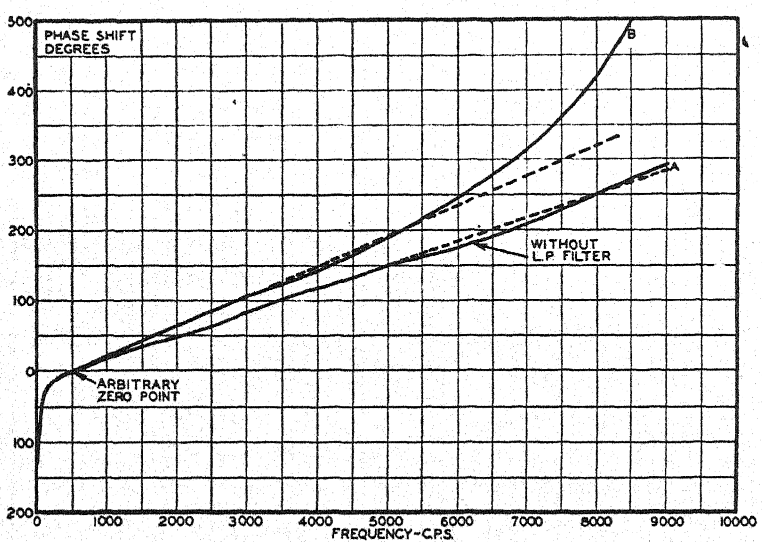


Figure 133 — Mixer + pre-amplifier + wide-range type amplifier + bridging amplifier + low-pass filter — phase shift curves.

Figure 133, Curve A, shows the phase shift characteristic of a series of amplifiers, the average value over the major portion of the frequency range being linear as indicated by the dotted line. In Curve B we see the curvature introduced by the addition of an m-derived low-pass filter of 7,500 cycles cut-off frequency. Note the deviation from linearity, also the increased slope indicating greater initial delay. In Figure 134 we see the frequency characteristic of this system, showing the sharp cut-off and rapid increase in attenuation toward the frequency of infinite attenuation.

The phase distortion data in all figures except Figures 126 and 127 were obtained from measurements made by the following method:

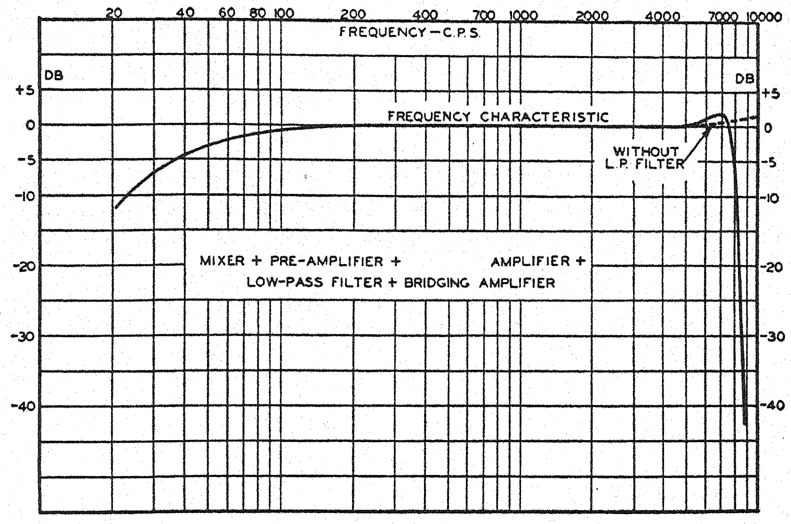


Figure 134 — Frequency characteristic of Figure 133.

Distortion delay data may be obtained from phase shift measurements, impedance measurements or direct measurements of envelope de-

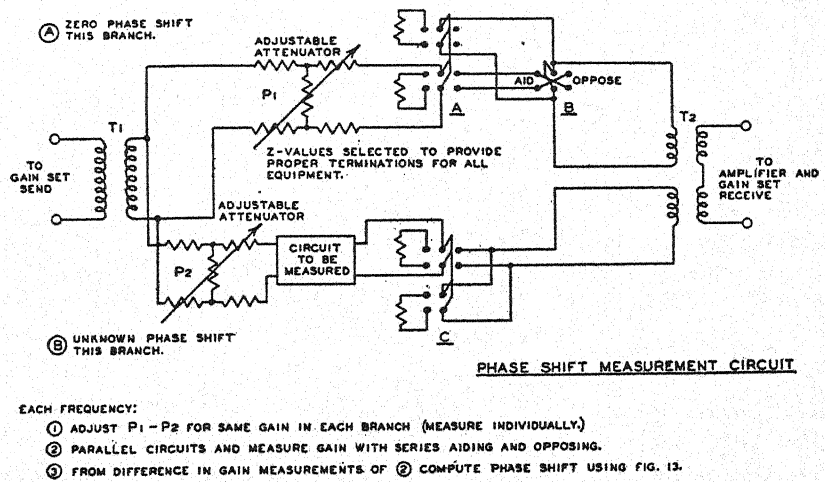


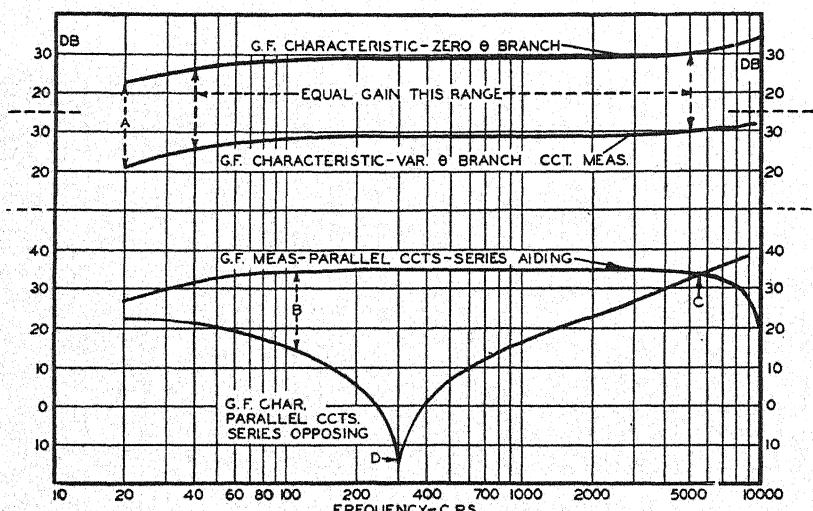
Figure 135 — Schematic diagram of phase shift measurement circuit.

lay. The phase shift method to be described is applicable to standard test equipment, will indicate small amounts of phase distortions, and is suffi-

ciently accurate for all practical purposes. This particular method of measuring phase shift is indicated schematically in Figure 135, and fundamentally consists of two parallel circuits, one with zero phase shift and the other with a varying and unknown phase shift. If, at any given frequency, we adjust the attenuators P_1 and P_2 so that the output currents at T_2 and consequently at the output of the common amplifier, are equal when the two circuits are used individually, we then have a simple condition of the vector sum and difference of the two equal currents when the circuits are paralleled and the switch B operated to series aiding or opposing. The difference between the vector sum and difference currents is indicated in db by the amount of attenuation necessary in the common circuit for equal output currents in this circuit in the "aid" and "oppose" position of the switch.

Summarized, the operations for measurement of phase shift, with reference to Figure 135, follow:

(1) Measure gain-frequency characteristic of branch B with branch A terminated.



(A) = Increased attenuation of zero β branch 1.5 db.
(B) = Difference in curves used with Figure 137 to determine phase shift.
(C) = Gain measurement the same for series aiding or opposing and phase shift equals 90°, 270°, 450°, etc., when vector sum equals vector difference.
(D) = Infinite attenuation in series opposing measurement when currents are 180° out-of-phase and phase shift equals 0°, 180°, 360°, etc.

Figure 136 — Measured curves of Figure 128 by circuit of Figure 135.

(2) Terminate circuit A and adjust attenuator P_1 so that the gain of circuit A over the flat portion of the curve is the same as the gain of

circuit B over a similar range, then proceed with gain-frequency measurement of circuit A.

(3) From these curves of A and B compute the variations in attenuation at $P_1 - P_2$, necessary to maintain equality in output currents (gain-frequency curves) in circuits A and B.

(4) Parallel circuits A and B and with switch B in "aid" position, measure gain-frequency characteristic of combination, adjusting attenuators as noted in (3), as required to maintain same transmission characteristics in both branches.

(5) Operate switch B to "oppose" and proceed with gain-frequency measurements of parallel circuits as in (4) [see note (1)].

(6) The resulting two gain-frequency curves will be similar to Figure 136, which is a measurement, made as outlined above, of the wide-range type recording amplifier of Figure 128. From these curves determine the difference in db between the aid and oppose condition at sufficient points to plot the phase shift in a degrees-frequency in c.p.s. curve using the chart in Figure 137 to convert db to degrees.

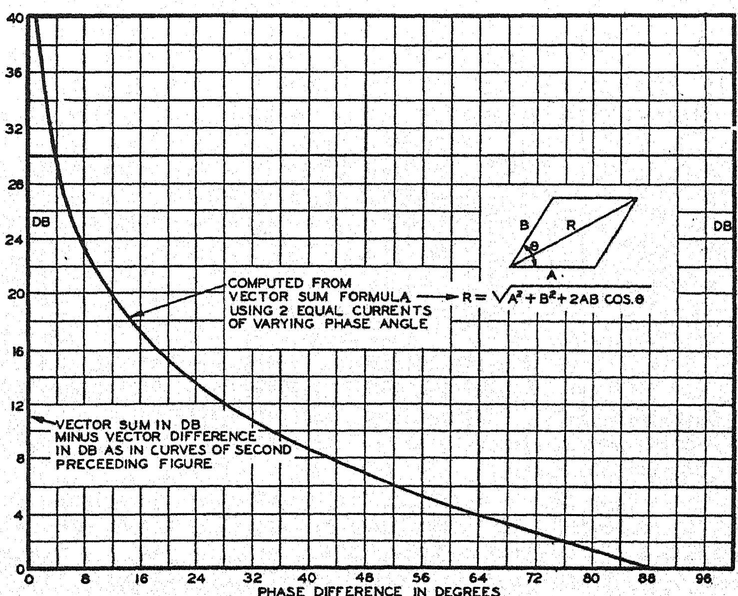


Figure 137 — Conversion chart-db/degrees for phase shift measurements.

NOTE 1:

An initial adjustment of the variable attenuators, with the circuits in the series opposing connection, and at the 0° — 180° etc., point, in

fractional db amounts, will facilitate attainment of sharp minimum values.

Again, in Figure 129 we show the resulting phase shift-frequency curve resulting from (6) in the preceding summary.

In Figure 138 note the frequency response of circuits A and B of Figure 135 when each circuit has "zero" phase shift. Note that the minimum difference in db is 44.5; referring to Figure 137, we see that this value corresponds to a phase shift of less than one degree, indicating that for practical measurements this particular circuit is satisfactory.

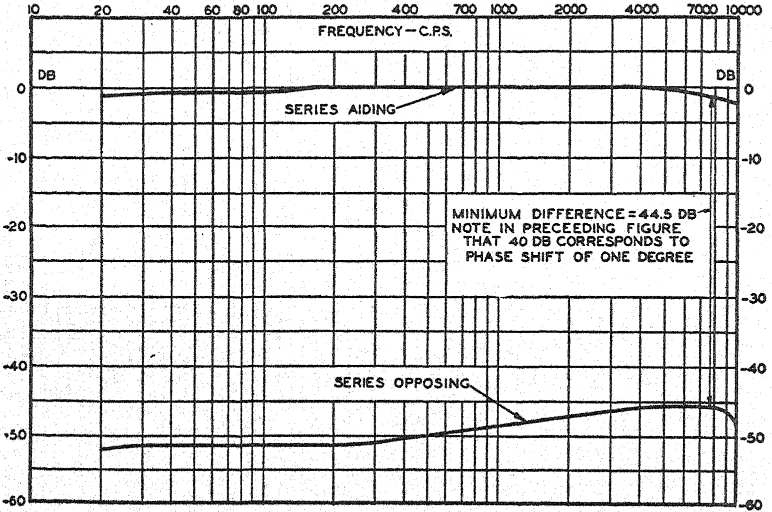


Figure 138 — Frequency response of Figure 135 with zero phase shift in each branch of parallel circuits.

While phase distortion is important in recording systems, it is not usually compensated for. If the frequency band is extended, as in television systems, or if a large number of amplifiers or filters are used in a circuit such as in a telephone system, such distortion becomes serious and must be deliberately corrected.

Correction of phase distortion is usually made by insertion of "all-pass" networks, having the property of phase shift without frequency attenuation. All-pass networks are described in detail in standard textbooks on transmission networks.

Acknowledgment is made to the *Bell System Technical Journal* for information and illustrations included in this chapter.

The above measurements were made by G. M. Sprague of Metro-Goldwyn-Mayer Studio Sound Department.

Chapter XIV

TRANSFORMERS FOR SOUND CIRCUITS

By JOHN K. HILLIARD

In power work, we are interested in changing or transforming voltages in a narrow low-frequency band from one value to another in order to effect economies in transmission, while in sound recording we are interested in transforming impedances from one value to another to obtain optimum efficiencies of transmission over a wide band of frequencies.

An impedance transforming device is known as an audio-frequency transformer, and in its ideal form modifies the magnitude of the load impedance to match it to the generator impedance. An audio-frequency transformer is similar to a power transformer except that it must operate over a wide band of frequencies instead of at a single frequency.

The ideal transformer neither stores nor dissipates energy, but in order to be ideal, a transformer would have to meet the following three requirements:

(1) Have infinite primary and secondary inductance, but a finite ratio of primary to secondary inductance.

(2) Have perfect coupling between primary and secondary.

(3) Have no resistance in primary and secondary windings.

In practice, these conditions are impossible to obtain, yet for the frequencies we use an approximation can be reached to the end that a transformer will transmit uniformly the band we require without appreciable loss:

The first condition is satisfied by using cores of extremely high permeability and a great number of turns on the coil;

The second condition is approximated by arranging the coils mechanically so that the leakage flux between turns is very small;

While the third condition is satisfied by the fact that for audio-frequencies the resistances of the windings are small compared to their impedance in the frequency ranges under consideration.

In an ideal transformer the ratio of primary to secondary voltage is equal to the ratio of the turns and hence the ratio of the currents is the

reciprocal of the voltage ratio, since there is no energy dissipated. To reflect the constants of the secondary side of the transformer to the primary, it is necessary to multiply the impedances of the primary by the ratio of secondary inductance to primary inductance or $\left(\frac{L_1}{L_2}\right)$.

For all practical purposes, with good iron core transformers the ratio of

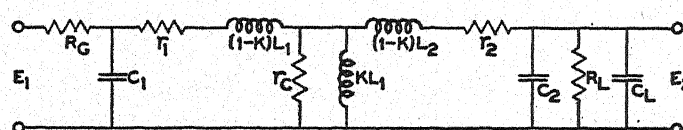
$$\frac{L_1}{L_2} = \left[\frac{N_1}{N_2}\right]^2 \tag{41}$$

also

$$\frac{E_1}{E_2} = \frac{N_1}{N_2} \text{ and } \frac{I_1}{I_2} = \frac{N_2}{N_1} \tag{42}$$

It is now possible to set up the formula for an equivalent circuit for an audio transformer, since in the practical transformer we have resistance in the coils which is the equivalent of a series loss, distributed capacities which represent shunt reactances on both primary and secondary, and losses in the magnetic circuit which look like resistance shunting the winding.

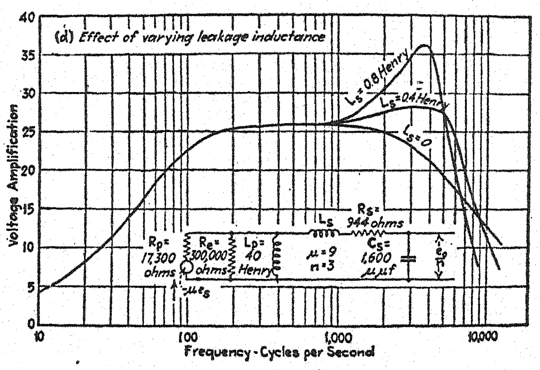
The circuit is as follows:



C_1 = distributed capacities of primary
C_2 = distributed capacities of secondary
C_L = distributed capacity of the load
K = coefficient of coupling
L_1 = inductance of primary
L_2 = inductance of secondary
r_1 = resistance of primary
r_2 = resistance of secondary
r_C = exciting current loss (core loss)
R_G = generator resistance
R_L = load resistance

Figure 139 — Equivalent circuit of audio transformer.

Since audio transformers are used to transmit a wide band of frequencies, we find that it is the extreme ends of the band which become the dominating factors. If we consider the transformer as acting between frequencies f_1 and f_2 as limits, then the mean frequency $f_M = \sqrt{f_1 f_2}$. We find that in the lower part of the band the leakage re-



—*From "Radio Engineering," by Frederick E. Terman (McGraw-Hill Book Co., Inc.).*

Figure 140.

actance $(1 - K)\ L_1$ (Figures 140 and 141), becomes very small as compared to the impedance of the primary. Also, the distributed capacities have a very high reactance as compared to the inductive reactance, and so the coil approximates the equivalent network of Figure 141.

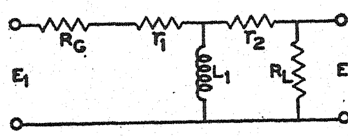


Figure 141 — Equivalent network of an audio-frequency transformer operating in the lower-frequency band.

And hence the ratio of

$$\frac{E_2}{E_1} = N \frac{R_L}{R_G + r_1 + r_2 + R_L + \dfrac{(R_G + r_1)\ (R_L + r_2)}{j\omega L_1}} \qquad (43)$$

At frequencies above the mean, the primary reactance (Figure 143) is so high that its shunting action is very small compared to the reactance of the distributed capacities, but at this point the leakage reactance must be considered as it starts to exert an appreciable effect.

This now gives the equivalent circuit of Figure 142.

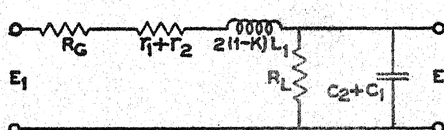
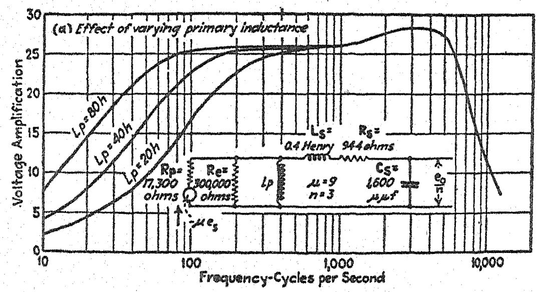


Figure 142 — Equivalent network of an audio-frequency transformer operating in the upper-frequency band.

Ordinarily an input transformer works into the grid of a vacuum tube, and the shunting effect of this high load impedance can be

neglected at the higher frequencies, so that the equivalent circuit be-



—From "Radio Engineering," by Frederick E. Terman (McGraw-Hill Book Co., Inc.).

Figure 143.

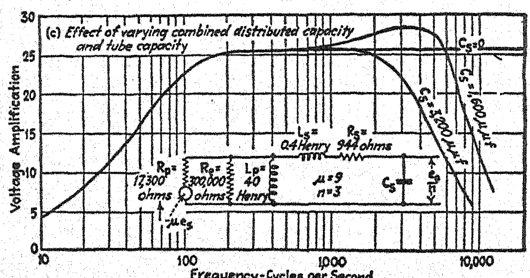
comes a series resonant circuit, and

$$\sqrt{L_e C} = \frac{1}{2\,\pi\, f_o} \tag{44}$$

where $L_e = 2\,(1 - K)\,L_1$

$f_o =$ the resonant frequency

From this equation we can determine the value of L and C.



—From "Radio Engineering," by Frederick E. Terman (McGraw-Hill Book Co., Inc.).

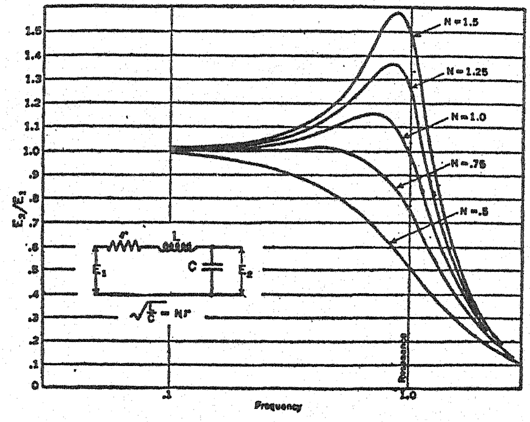
Figure 144.

If the amplification is to be uniform at high frequencies, f_o must be past the limit of the band. In input transformers it is desirable to obtain the maximum possible turn ratio, which increases both L_e and C. Hence, both quantities must be kept very low. In order to keep the amplitude at resonance low, resistance wire may be used to wind the secondary, or a very high resistance may be inserted in the circuit.

Since leakage reactance varies with N^2, and the turns ratio must be

high, there are two conflicting demands, one for many turns and the other for few.

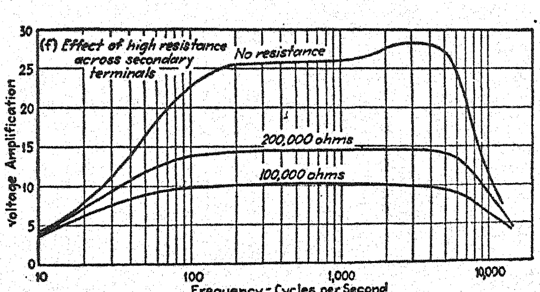
The output transformer differs from the input and inter-stage transformer in that it is a stepdown transformer, and the impedances between which it works are usually considerably less, which makes



—*Reprinted by permission from "Electrical Engineers' Handbook," by Pender and McIlwain, published by John Wiley & Sons, Inc.*

Figure 145 — Curves showing leakage resonance for different ratios of inductance, capacity, and resistance.

possible the use of a proportionately large number of turns without introducing the inherent difficulties of high leakage reactances and distributed capacity. Consequently, if weight is of no particular importance, the core material may be of the lower permeability grade (Figure 148).

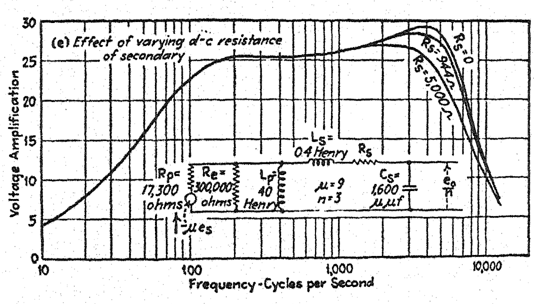


—*From "Radio Engineering," by Frederick E. Terman (McGraw-Hill Book Co., Inc.).*

Figure 146.

Improvements in frequency response during the past few years have required marked changes in transformers, some of which have

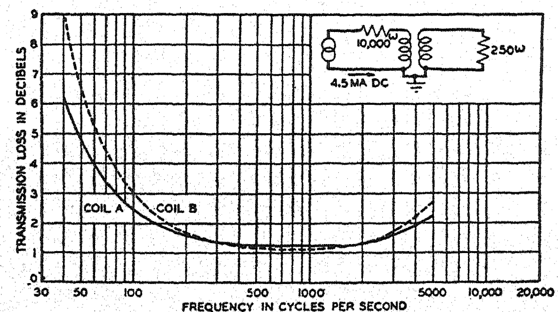
been in the application of new magnetic core materials, such as permalloy and hypernik (iron-nickel alloys having from 40-90% nickel).



—*From "Radio Engineering," by Frederick E. Terman (McGraw-Hill Book Co., Inc.).*

Figure 147.

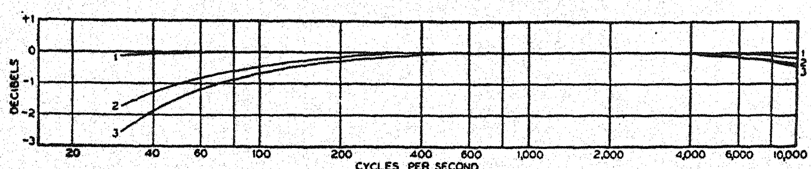
As a result, extremely high permeabilities are obtained with very small weight and loss distortion over wider frequency bands (Figure



—*From the Bell System Technical Journal.*

Figure 148 — Curves showing the transmission loss of (A) small output transformer, and (B) larger output transformer.

149). Reduced phase distortion is obtained since the inductance of the windings is much greater than that required for the proper trans-



Curves showing the effect of different materials on the frequency response of an audio transformer

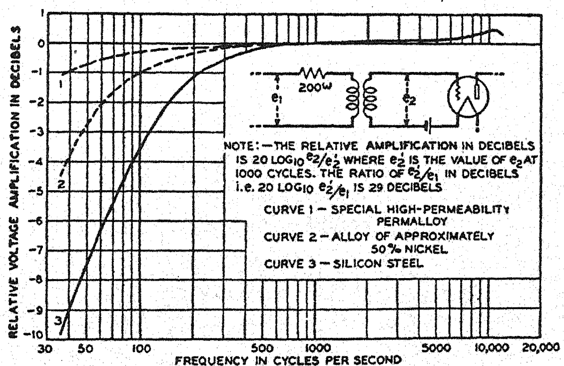
1—Alloy containing 45-50 per cent nickel 2—Experimental iron 3—Transformer steel, audio grade

—*From a technical paper in Electrical Engineering, Volume 54, 1935, "Silicon Steel and Communication Equipment," by C. H. Crawford and E. J. Thomas of the General Electric Company.*

Figure 149.

mission loss. Also, in most transformers, a certain amount of magnetic modulation takes place due to a magnetic non-linearity, usually indicated by harmonics of the lower frequencies occurring in the high range. This energy at the low frequencies is very great, as compared to that at very high frequencies, and for this reason, the harmonics of the low frequencies may be of the same order as the high-frequency signal.

The audio-frequency transformer differs from the power transformer in that the latter operates from a voltage source of good regulation. The audio transformer, however, operates from a voltage source of poor regulation, varying widely in amplitude and frequency range. As a result, the flux densities vary from negligible values at 10,000 cycles to approximately 5,000 lines per square centimeter at 30 cycles. Under these conditions, it is necessary to provide sufficient impedance in order that a constant voltage amplitude ratio be maintained over the operating range. High-permeability steel permits small core size, which



—*From the Bell System Technical Journal.*

Figure 150 — Showing relative effect of various core materials on the transmission characteristics of portable recording apparatus.

in turn gives lower distributed capacity and lower leakage inductance, both of which depend upon the geometry of the coil.

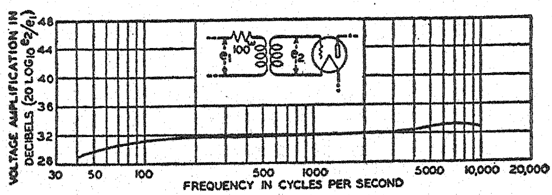
Since this is not detected in ordinary measurements, harmonic analyses must be resorted to.

The most efficient use of filters and equalizers requires constant impedance inputs or terminations. Better impedance characteristics of transformers are obtained by increasing the mutual impedance, and decreasing the leakage and capacity reactance variations.

In measuring the performance of a transformer, care must be taken to make sure that the conditions under which it is to operate are reproduced in the testing circuit, and precautions must be observed in the

tests in order that a misleading performance characteristic will not be obtained.

The permeability of magnetic core materials tends to rise rapidly from its initial values with increasing current. For this reason, if tests

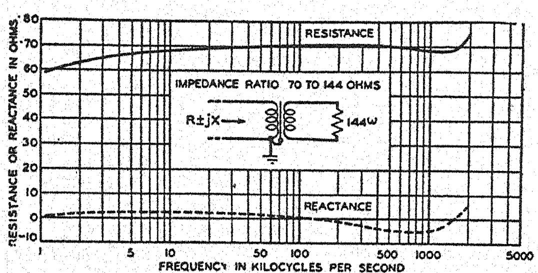


—*From the Bell System Technical Journal.*

Figure 151 — Transmission characteristic of light weight input transformer having a weight of 3½ ounces.

were made with low frequencies at considerably higher currents than are obtained under the operating condition, the measured low-frequency characteristic would indicate a much better response than that which would be obtained under service conditions.

Considerable demand has been created for portable apparatus, both for recording and testing, and as a result, we now have extremely light weight transformers of very small size. The gain-frequency characteristic, as shown in Figure 151, indicates that it is possible to construct these small transformers for specific uses with efficiencies which are comparable to, or better than, the larger type transformer.



—*From the Bell System Technical Journal.*

Figure 152 — Impedance-frequency characteristic of a well designed transformer.

Transformer Problems

1. *Given:*

   Transformer step-up ratio of 1 to 5

   $Z_1 = 500$ ohms

   $I_1 = 0.01$ amps.

   $E_1 = 10$ volts

   $K =$ constant of coupling $= 1$

   Find:

   Z_2, I_2, and E_2

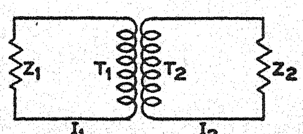


Figure 153 — Circuit diagram for transformer problems.

Circuit diagram (Figure 153):

$$\frac{Z_1}{Z_2}=\frac{N_1^2}{N_2^2};\quad \frac{500}{Z_2}=\frac{1}{25};\quad Z_2=500\times 25=12{,}500 \text{ ohms}$$

$$\frac{I_1}{I_2}=\frac{N_2}{N_1};\quad \frac{0.01}{I_2}=\frac{5}{1};\quad I_2=\frac{0.01}{5}=0.002 \text{ amps}$$

$$\frac{E_1}{E_2}=\frac{N_1}{N_2};\quad \frac{10}{E_2}=\frac{1}{5};\quad E_2=5\times 10=50 \text{ volts}$$

2. If the input impedance to a transformer is 200 ohms, and the load impedance is 20,000 ohms, what is the turns ratio of the transformer?

$$\frac{Z_1}{Z_2}=\frac{N_1^2}{N_2^2};\quad \frac{N_1}{N_2}=T \text{ (turns ratio)}$$

$$T^2=\frac{Z_1}{Z_2}=\frac{200}{20{,}000}=\frac{1}{100}$$

$$T=\frac{1}{10}$$

3. If the plate resistance of a vacuum tube is 10,000 ohms, what must be the impedance and inductance of the primary of a transformer, coupled directly to the tube, in order that the tube output will be attenuated 3 db at 30 cycles?

$$20\log\frac{E_G}{E_L}=3 \text{ db* (45)}$$

$$\log\frac{E_G}{E_L}=0.15$$

$$\frac{E_G}{E_L}=1.413$$

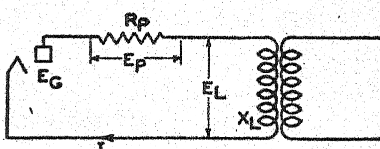


Figure 154 — Circuit diagram for problem 3.

$$\text{so}\quad \frac{E_L}{E_G}=0.707$$

Where E_L = voltage across the load

E_G = theoretical output voltage of the tube (considering the plate to filament resistance as external to the tube)

* [Formula (45) can be found directly from the Table, Chapter XXXII.]

$$\frac{E_L}{E_G} = 0.707 = \sin 45°$$

$$\therefore \quad \text{phase angle} = 45°$$

$$\tan 45° = 1 = \frac{E_L}{E_P}$$

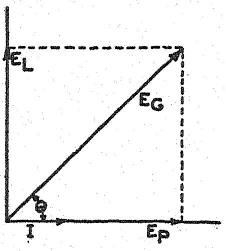


Figure 155 — Vector diagram

and
$$\frac{E_L}{X_L} = \frac{E_P}{R_P} \text{ so } \frac{X_L}{R_P} = 1 \text{ and } X_L = R_P$$

$$X_L = 10{,}000 = 2\,\pi f L = 2\,\pi \times 30 \times L$$

$$L = \frac{10{,}000}{2\,\pi \times 30} = \frac{10{,}000}{188.4}$$

$$L = 53 \text{ henries}$$

Chapter XV

GENERAL NETWORK THEORY

By HARRY KIMBALL

1. NATURE OF NETWORKS

In the several fields of electrical engineering the term "transmission network" is used rather loosely to refer to a variety of types of electrical circuits. For instance, to the power engineer, a transmission network may mean an interconnected arrangement of transmission lines employed for the purpose of transmitting electrical power from point to point. Another use of the term, as derived from the communication field, is in reference to circuits which can be obtained from the connecting together of resistances, condensers, and inductance coils including transformers, to form equipments for use primarily as links of systems, whose function is to transmit electrically, acoustic or other signals conveying intelligence. These networks necessarily have four terminals, two for connecting to a transmission system to receive power, and two for connecting to a load to deliver power. For this reason, they are sometimes known as "four-terminal networks." The resistances, condensers, and inductance coils contained within such a network are known as the "network elements" and their electrical values are termed the "network constants."

It will be seen later that the internal structure of a network naturally divides itself into groups of electrical elements which form the branches or arms of the network. In the limiting case, these groups, may, and often do, consist of only one element. It is usually simpler in working out the mathematical relations of a network to deal with these natural groups as units rather than the individual elements. Such groups, whether consisting of one or more elements, require two accessible terminals for connecting them into the network structure. For this reason they are often known as "two-terminal networks," that is, two-terminal networks are the groups of elements from which four-terminal networks are made. A knowledge of the impedance characteristics of two-terminal networks is an important prerequisite in studying four-terminal networks. Certain fundamental information concerning these networks is therefore presented in the following work prior to the discussion of four-terminal networks.

For the networks which may be constructed in the above manner, there exist several well defined types designated in accordance with the duties they perform when inserted into a transmission facility. These networks are attenuators, attenuation equalizers, electric wave filters, and phase correctors or delay networks defined in the following material.

ATTENUATORS—An attenuator is a network whose insertion loss remains constant with change of frequency. Departure from constant attenuation occurs only when the design limitations of the component electrical elements are exceeded. The design and use of attenuators is discussed in Chapter XXXIII.

ATTENUATION EQUALIZERS—An attenuation equalizer is a network whose insertion loss varies in some desirable manner with change of frequency. Such networks usually are inserted into transmission systems to compensate for defects in the transmission-frequency characteristics of the systems. These networks are discussed in detail in the following chapter.

ELECTRIC WAVE FILTERS—An electric wave filter is a network which transmits without appreciable attenuation all frequencies within one or more frequency bands and attenuates all other frequencies. In practice, very few wave filters have more than one transmission band. Wave filters are discussed in a following chapter.

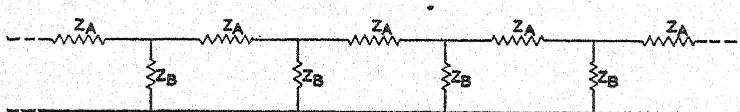


Figure 156-A — Ladder-type circuit.

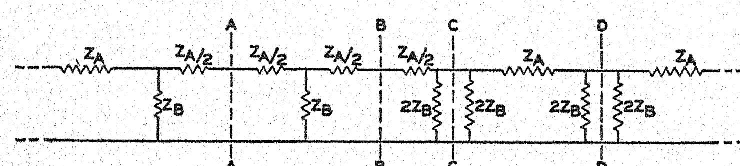


Figure 156-B — Ladder-type circuit.

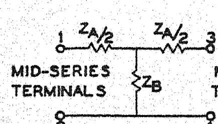


Figure 156-C — T type section.

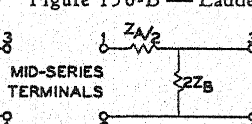


Figure 156-D — L type section.

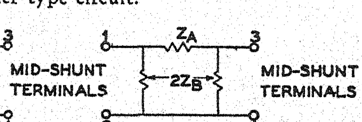


Figure 156-E — π type section.

Ladder-type sections.

PHASE CORRECTORS—A phase corrector is a network having the property of changing in a preassigned manner the relative phase rela-

tion of the frequencies of a given transmission band. Except for the unavoidable dissipation of the electrical elements, these networks do not attenuate the frequencies of their transmission range. Phase correctors are used with transmission systems to correct for system phase distortion similar to the manner in which attenuation equalizers correct for attenuation distortion. These networks are not within the scope of this material. (See Chapter XIII.)

It has been found that the engineering of networks is considerably facilitated by the use of distinctive unit structures operated in tandem to form configurations which, in many respects, resemble a carpenter's ladder. Many of the symbols and conceptions used in network theory thus have their origin in these "ladder-type" networks. For instance, referring to Figure 156-A, the circuit shown is a ladder-type structure having series impedances denoted by the symbol Z_A and shunt impedances denoted by Z_B. If we arrange this circuit as shown in Figure 156-B, and remove a portion of the network by cutting through two adjacent series arms at their mid-points, as shown by "AA" and "BB," we obtain the familiar T type section of Figure 156-C. The reason then for designating the series arms of this section by the symbol $Z_A/2$ and the shunt arm by the symbol Z_B is thus made clear. Also, the origin of the term "mid-series terminals" when referring to either of the pairs of terminals of a T section is evident. In a similar way, if we remove another portion of the ladder-type network by cutting through two adjacent shunt arms at their mid-points in the manner shown by "CC" and "DD," we obtain the π type section of Figure 156-E. Following the same method of designation, the series arm impedance of this section is denoted by the symbol Z_A, the shunt arms by the symbol $2\,Z_B$, and the two pairs of terminals are called "mid-shunt" terminals. Another familiar ladder-type section which can be obtained by removing a portion of the general ladder structure is the L type section of Figure 156-D. This section has a series arm of $Z_A/2$ and a shunt arm of $2\,Z_B$. The 1-2 terminals of the L section of Figure 156-D are the mid-series terminals and the 3-4 terminals are the mid-shunt terminals. L type sections are often referred to as half sections for the reason that two L sections connected together form one T or one π type section depending upon whether they are joined at their mid-shunt or mid-series terminals.

Transmission systems employing networks usually are arranged so that the impedances connected to the input and output terminals of the networks are resistances. That is, the impedance looking back into the system at the network input terminals is chiefly resistive in character as is also the load impedance connected to the output terminals.

The symbol R_G is used to designate the sending-end system impedance connected to a network and R_L denotes the system load impedance. Where R_G and R_L are equal they are designated in the following work by the symbol R_0. For proper performance, wave filters require that R_G and R_L be equal resistances unless the filters have impedance transforming designs. In the case of attenuation equalizers, R_G and R_L need not be equal resistances for proper performance although they are usually arranged to be equal.

For each of the types of four-terminal networks mentioned above there exists an orderly procedure of design made available as the result of a vast amount of theoretical work coupled with practical experience. These design procedures are, of course, interrelated, the methods employed for one type of network having much in common with those used for other types. It is desirable in taking up the study of networks to consider first a certain amount of "ground material" useful for any type of network. The remainder of the material in this chapter is mostly of this nature.

2. TWO-TERMINAL REACTIVE NETWORKS

Certain two-terminal reactive networks, that is, circuits not containing resistance elements, are used so frequently in the design of equalizers and filters that it is helpful to arrange the formulae for their reactances into forms especially suited to this type of work. Usually the reactance equations of such networks are expressed in terms of the electrical constants of the network elements. When the network consists of, say N elements, N electrical constants appear in the corresponding reactance equation. In equalizer and filter work it is convenient to eliminate the electrical constants from the reactance equations and use, in their place, the important critical frequencies of the networks; that is, the frequencies of resonance and anti-resonance. For any reactive network, however, the sum of the number of resonant and anti-resonant points is one less than the number of network elements. For instance, for a four-element reactive network, the resonant and anti-resonant points will number three. Hence, if these critical frequencies are used in the formulae to replace the electrical constants we are short one item for substitution purposes. To obviate this difficulty another critical frequency designated as f_S is used. The definition of f_S will be made clear as we progress. Proper selection of this frequency in connection with the equalizer work of the next chapter has made possible the simple resulting equations for the insertion losses of the networks. Following is shown the method for expressing the reactances of some simple two-terminal reactive networks in terms of these new parameters.

(a) Reactance of an Inductance Coil

The vector reactance of an inductance coil of L henries is usually expressed as follows:

$$jX = j\,2\,\pi\,f\,L. \tag{46}$$

This simple reactance network has, of course, no finite resonant or anti-resonant frequencies and hence these parameters cannot be used for substitution purposes. As a means of eliminating L from this equation, define f_s as the frequency where the reactance is, say X_s. Then we have

$$X_s = 2\,\pi\,f_s\,L$$

Solving for L gives:

$$L = \frac{X_s}{2\,\pi\,f_s} = \frac{X_s}{\omega_s} \text{ henries} \tag{47}$$

Equation (46) can now be written

$$jX = jX_s\,\frac{f}{f_s} \tag{48}$$

We have thus expressed the reactance of the inductance coil in terms of a frequency ratio f/f_s and the reactance X_s. To illustrate, suppose we wanted an inductance coil having a reactance of 500 ohms at 3000 c.p.s. From equation (47)

$$L = \frac{X_s}{\omega_s} = \frac{500}{2\pi \times 3000} = 0.0265 \text{ henries}$$

And the reactance of the coil at any frequency, f, is

$$jX = j\,500\,\frac{f}{3000}$$

(b) Reactance of a Condenser

In a similar manner the reactance of a condenser can be expressed in terms of a reactance X_s and a frequency ratio. The usual equation for the reactance of a condenser is

$$-jX = -j\,\frac{1}{2\pi\,f\,C} \tag{49}$$

At a frequency f_s let the numerical value of the reactance be X_s. Then

$$X_s = \frac{1}{2\pi\,f_s\,C}$$

Solving for C gives

$$C = \frac{1}{2\pi\,f_s\,X_s} = \frac{1}{\omega_s\,X_s} \text{ farads} \tag{50}$$

Using this value of C with equation (49) gives

$$-jX = -jX_S \frac{f_S}{f} \tag{51}$$

As an example, suppose we wanted a condenser having a reactance of 500 ohms at 3000 c.p.s. Then from (50)

$$C = \frac{1,000,000}{2\pi \times 3,000 \times 500} = 0.106 \text{ microfarads}$$

and from equation (51)

$$-jX = -j\,500\,\frac{3,000}{f}$$

(c) Reactance of a Coil and Condenser in Series

In this case let Figure 157-A represent a two-terminal network consisting of an inductance L in series with a capacitance C. Figure 157-B shows symbolically the reactance characteristic of such a circuit

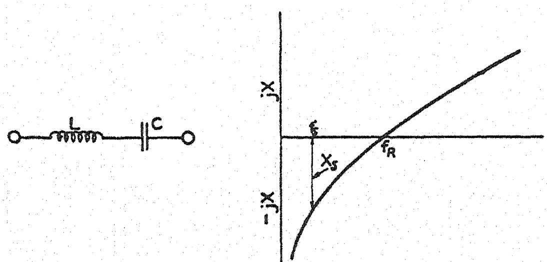


Figure 157-A — Inductance coil and condenser in series. Figure 157-B — Reactance characteristic.

plotted against frequency. The symbol f_R is used to denote the resonant frequency of the coil and condenser; that is

$$f_R = \frac{1}{2\pi\sqrt{LC}} \tag{52}$$

Again, as in the previous cases, f_S is defined as the frequency where the reactance of the circuit is numerically equal to X_S. Since this happens at two points in the frequency range, that is, once above the resonant frequency f_R and again below f_R, f_S is taken as the lower point as shown in Figure 157-B. The symbol "s" is used to denote the ratio of f_R to f_S, that is

$$s = \frac{f_R}{f_S} \text{ (greater than unity)}$$

The reactance equation for the circuit is

$$jX = j\left(\omega L - \frac{1}{\omega C}\right)$$

$$jX = j\omega_R L\left[\frac{\omega}{\omega_R} - \frac{1}{\omega\omega_R LC}\right]$$

But from equation (52) $(\omega^2_R) = \frac{1}{LC}$

and we have

$$jX = j\omega_R L \left[\frac{\omega}{\omega_R} - \frac{\omega_R}{\omega}\right]$$

$$jX = j\omega_R L \left[\frac{f}{f_R} - \frac{f_R}{f}\right] \tag{53}$$

by definition when $f = f_S$ then $X = -X_S$ (see Figure 157-B) which gives

$$\omega_R L = \frac{-X_S}{\frac{f_S}{f_R} - \frac{f_R}{f_S}} = \frac{X_S}{s - \frac{1}{s}} \tag{54}$$

Use of this value of $\omega_R L$ in connection with equation (53) gives us the form which we desire as a means of expressing the above reactance; that is,

$$jX = jX_S \frac{\frac{f}{f_R} - \frac{f_R}{f}}{s - \frac{1}{s}} \tag{55}$$

The values of L and C can also be expressed in terms of the new parameters in the following manner, by means of equation (54) and the relation

$$\omega_R = s\,\omega_s$$

We have

$$L = \frac{X_S}{\omega_R}\,\frac{s}{s^2 - 1} = \frac{X_S}{\omega_S}\,\frac{1}{s^2 - 1} \tag{56}$$

And when this value of L is used with equation (52) we have for C

$$C = \frac{1}{\omega_R{}^2 L} = \frac{1}{\omega_R X_S}\,\frac{s^2 - 1}{s} = \frac{1}{\omega_S X_S}\,\frac{s^2 - 1}{s^2} \tag{57}$$

As an illustration of the above, consider a series coil and condenser circuit having its resonant frequency at 5,000 c.p.s. and a reactive impedance of 500 ohms at 4,000 c.p.s. That is,

$$f_R = 5{,}000 \text{ c.p.s.} \qquad s = 1.25$$
$$f_S = 4{,}000 \text{ c.p.s.} \qquad X_S = 500 \text{ ohms.}$$

This gives

$$L = \frac{500}{2\,\pi\, 4{,}000}\,\frac{1}{(1.25)^2 - 1} = 0.0353 \text{ henries}$$

$$C = \frac{1{,}000{,}000}{2\,\pi \times 4{,}000 \times 500}\,\frac{(1.25)^2 - 1}{(1.25)^2} = 0.0287\ \mu f$$

$$jX = j\,\frac{500 \times 1.25}{(1.25)^2 - 1}\left[\frac{f}{f_R} - \frac{f_R}{f}\right] = j\,1{,}111\left[\frac{f}{f_R} - \frac{f_R}{f}\right]$$

(d) Reactance of a Coil and Condenser in Parallel

Figure 158-A represents a two-terminal network consisting of a coil and condenser in parallel and Figure 158-B shows the form of the reactance characteristic. The symbol f_R designates the frequency of

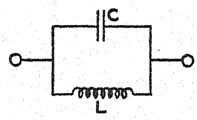


Figure 158-A — Inductance coil and condenser in parallel.

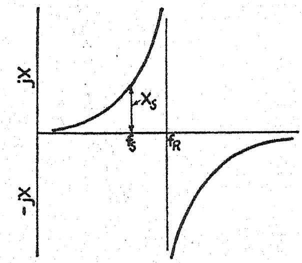


Figure 158-B — Reactance characteristic.

anti-resonance and the symbol f_S is used to indicate the point where a reactance of numerical value X_S is obtained. The reactance of this network is usually written in the form

$$jX = \frac{\frac{\omega L}{\omega C}}{j\left(\omega L - \frac{1}{\omega C}\right)}$$

which may be converted into the form

$$jX = \frac{\frac{\omega L}{\omega C}}{j\omega_R L\left[\frac{\omega}{\omega_R} - \frac{1}{\omega\omega_R}\,\frac{1}{LC}\right]}$$

At the resonant frequency, $\frac{1}{LC} = \omega^2{}_R$. Substituting this value of LC gives

$$jX = \frac{1}{j\omega_R C}\left[\frac{1}{\frac{\omega}{\omega_R} - \frac{\omega_R}{\omega}}\right] = \frac{1}{j\omega_R C}\left[\frac{1}{\frac{f}{f_R} - \frac{f_R}{f}}\right] \qquad (58)$$

Now by definition when $f = f_S$; $X = X_S$, which gives

$$\frac{1}{j\omega_R C} = jX_S\left[\frac{f_S}{f_R} - \frac{f_R}{f_S}\right] = jX_S\left[\frac{1}{s} - s\right]$$

$$\frac{1}{j\omega_R C} = -\,jX_S\left[s - \frac{1}{s}\right] \qquad (59)$$

Use of this value in connection with equation (58) gives the final form we desire for expressing the reactance of the parallel coil and condenser.

$$jX = -jX_S \frac{s - \frac{1}{s}}{\frac{f}{f_R} - \frac{f_R}{f}} \tag{60}$$

The values of L and C in terms of the new parameters may now be determined from (59)

$$C = \frac{1}{\omega_R X_S} \frac{1}{s - \frac{1}{s}} = \frac{1}{\omega_S X_S} \frac{1}{s^2 - 1} \tag{61}$$

And since at the resonant frequency, $\omega_R L = \frac{1}{\omega_R C}$ we have for L

$$L = \frac{1}{\omega^2_R C} = \frac{\omega_R X_S}{\omega^2_R}\left(s - \frac{1}{s}\right) = \frac{X_S}{\omega_R}\left(s - \frac{1}{s}\right)$$

$$L = \frac{X_S}{\omega_S} \frac{s^2 - 1}{s^2} \tag{62}$$

3. EQUIVALENT TWO-TERMINAL NETWORKS

The impedance of a two-terminal network is, of course, that obtained looking in at the terminals and takes into account the effects of all of the network elements. It is possible for certain pairs of such networks to have identical impedances at their terminals for any frequency and yet have different circuit arrangements and different element values. A pair of such networks, however, must each have the same number of elements with a definite relationship existing between corresponding elements. Two networks of these types are said to be equivalent. Not all two-terminal networks have equivalent networks. In general, however, any two-terminal network having two or more elements of the same kind has an equivalent network. By this is meant two or more resistance elements, two or more inductance elements, etc. Since equivalent networks display identical impedances at their terminals, they may be used interchangeably as parts of four-terminal networks. In this connection they are often of great commercial value for it frequently happens that one of the networks is easier and cheaper to construct than the other.

Single- and two-element networks have no equivalent circuits except those that are exactly identical, and these are not regarded as

being equivalent arrangements. For networks containing three electrical elements, important equivalent networks exist if the network has two elements of like kind. When the three elements of such a network are all different in kind, that is, a resistance, a capacitance and an inductance coil, no equivalent network exists. Two-terminal networks containing four or more elements also have equivalent networks but are not discussed in this book. However, the conditions for the equivalence of a pair of three-element networks are discussed in detail.

To illustrate the above, consider circuits A and B of Figure 159, where R_1 and R_2 are any two resistances and n is any number. The

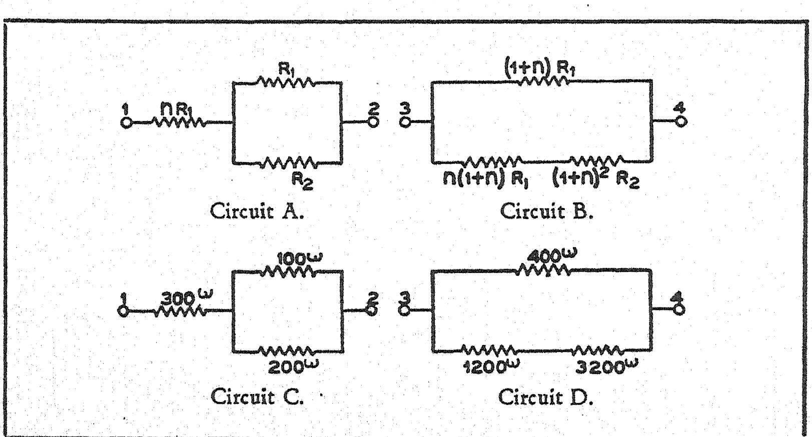


Figure 159 — Equivalent resistive networks.

resistance obtained at the terminals of circuit A of Figure 159 is

$$R_{12} = nR_1 + \frac{R_1 R_2}{R_1 + R_2} = \frac{nR^2_1 + R_1 R_2 (1 + n)}{R_1 + R_2} \qquad (63)$$

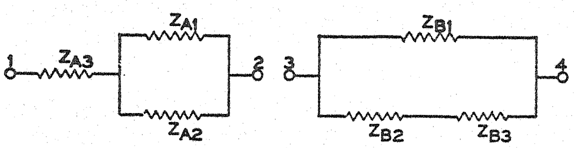
The resistance of circuit B of Figure 159 is

$$R_{34} = \frac{(1 + n)\ R_1\ [n\ (1 + n)\ R_1 + (1 + n)^2 R_2]}{(1 + n)\ \ [R_1 + n\,R_1 + (1 + n)\ R_2]}$$

$$R_{34} = \frac{n\,R^2_1 + (1 + n)\ R_1 R_2}{R_1 + R_2} \qquad (64)$$

The resistances of the two networks are thus shown to be identical for any values of R_1, R_2, and n. Suppose, for instance, that $R_1 = 100$ ohms, $R_2 = 200$ ohms, and $n = 3$. This results in the two specific resistive circuits, C and D, of Figure 159, each of which may be shown to present a resistance of 367 ohms at its terminals.

The above illustration uses resistances for the elements of the two equivalent networks. The equivalence of a pair of similar networks



Circuit A Circuit B

IMPEDANCE RELATIONS

$$n = \frac{Z_{A3}}{Z_{A1}} = \frac{Z_{B3}}{Z_{B1}}$$

$$\frac{Z_{B1}}{Z_{A1}} = (1 + n) \qquad \frac{Z_{B2}}{Z_{A2}} = (1 + n)^2 \qquad \frac{Z_{B3}}{Z_{A3}} = (1 + n)$$

Figure 160 — Equivalent three-element two-terminal networks.

can also be shown where impedances are used instead of resistances. Consider circuit A and circuit B of Figure 160. The impedances of these two networks are identical when their respective electrical elements are related in the manner shown by the equations given on the figure. This can be demonstrated as follows: Considering circuit A, its impedance is

$$Z_{12} = Z_{A3} + \frac{Z_{A1} Z_{A2}}{Z_{A1} + Z_{A2}} = \frac{Z_{A1} Z_{A2} + Z_{A1} Z_{A3} + Z_{A2} Z_{A3}}{Z_{A1} + Z_{A2}} \qquad (65)$$

And for circuit B we have

$$Z_{34} = \frac{Z_{B1} Z_{B2} + Z_{B1} Z_{B3}}{Z_{B1} + Z_{B2} + Z_{B3}}$$

Using the relation of elements shown in Figure 160 we may write

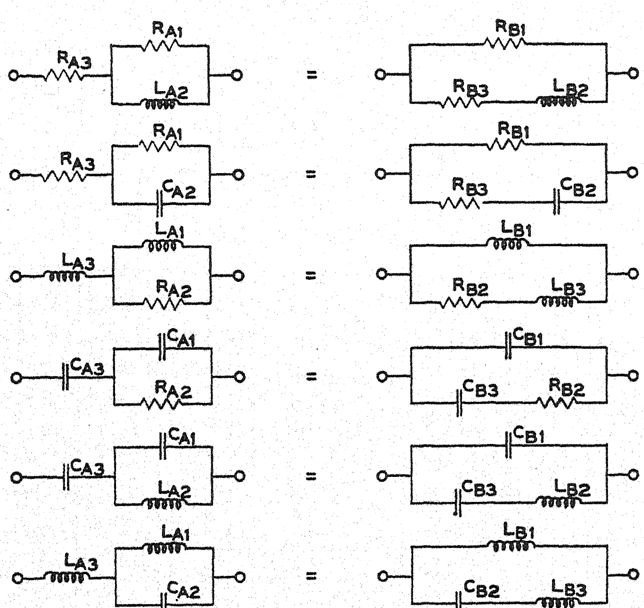
$$Z_{34} = \frac{Z_{A1} Z_{A2} (1 + n)^3 + Z_{A1} Z_{A3} (1 + n)^2}{Z_{A1} (1 + n) + Z_{A3} (1 + n) + Z_{A2} (1 + n)^2}$$

$$Z_{34} = \frac{(1 + n) [Z_{A1} Z_{A2} + nZ_{A1} Z_{A2} + Z_{A1} Z_{A3}]}{Z_{A1} + nZ_{A1} + Z_{A2}(1 + n)}$$

$$Z_{34} = \frac{Z_{A1} Z_{A2} + Z_{A1} Z_{A3} + Z_{A2} Z_{A3}}{Z_{A1} + Z_{A2}}$$

$$Z_{34} = Z_{12} \qquad (66)$$

Hence, for the impedance relations shown, the two circuits A and B are equivalent. It is noted that for the practical use of these equivalent circuits the ratio n must be a real number. This is only realized when Z_{A1} and Z_{A3}, or Z_{B3} and Z_{B1} are the same kind of circuit elements; that is, both resistances, or both inductances, etc. The impedance equations of Figure 160 have been set up in the manner shown so that we can readily obtain circuit B when circuit A is known, or conversely. Figure 160 shows a number of pairs of three-element equivalent networks obtained by assigning specific element types to the general impedances of Figure 160.



$$n = \frac{R_{A3}}{R_{A1}} = \frac{R_{B3}}{R_{B1}} = \frac{L_{A3}}{L_{A1}} = \frac{L_{B3}}{L_{B1}} = \frac{C_{A1}}{C_{A3}} = \frac{C_{B1}}{C_{B3}}$$

$$(1+n) = \frac{R_{B3}}{R_{A3}} = \frac{L_{B3}}{L_{A3}} = \frac{C_{A3}}{C_{B3}} = \frac{R_{B1}}{R_{A1}} = \frac{L_{B1}}{L_{A1}} = \frac{C_{A1}}{C_{B1}}$$

$$(1+n)^2 = \frac{R_{B2}}{R_{A2}} = \frac{L_{B2}}{L_{A2}} = \frac{C_{A2}}{C_{B2}}$$

Figure 161 — Pairs of three-element equivalent networks.

4. INVERSE TWO-TERMINAL NETWORKS

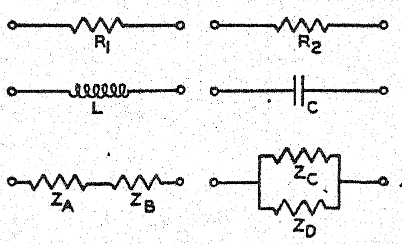
The theory of inverse networks relates to an important mutual relationship which may be made to exist between the impedances of a pair of two-terminal networks when the electrical constants and configuration of one network are properly selected with respect to the other network. Consider a pair of two-terminal networks one of which presents an impedance Z_1 at its terminals, and the other having an impedance Z_2. When Z_1 and Z_2 are so related that

$$Z_1 Z_2 = R_0^2 \tag{67}$$

the networks are defined as being the reciprocal or inverse of each other with respect to R_0. In general R_0 may be a vector expression but in equalizer and filter work it is convenient to let R_0 be the resistive impedance between which the networks operate: In other words, R_0 is a real number and a known quantity in most network problems. It is noted that reciprocal or inverse networks always involve a pair of networks mutually related to each other. Complete information must be available concerning the electrical elements and circuit connections of one network before its inverse network can be found. Also, the inverse network has not been completely determined until its electrical constants and the manner in which they are connected together are known.

The inverse network corresponding to a given network whose constants and connections are known may be derived with the aid of the few pairs of simple inverse networks given in Figure 162.

PAIRS OF INVERSE NETWORKS



RELATION BETWEEN CIRCUIT CONSTANTS	RULE
$R_1 R_2 = R_0^2$	(1)
$L/C = R_0^2$	(2)
$Z_A Z_C = Z_B Z_D = R_0^2$.	(3)

Figure 162 — Fundamental pairs of inverse networks.

The relationships shown between the electrical constants for the above pairs of inverse networks have been determined on the basis that their

corresponding impedances satisfy equation (67). It is left for the student to verify this to his own satisfaction. It is noted that:

(1) When one of a pair of inverse networks is a resistance the other network is another resistance.

(2) When one of a pair of inverse networks is an inductance the other is a capacitance, and conversely.

(3) When one of a pair of inverse networks consists of two impedances in series, the other consists of the inverse of these impedances in parallel, and conversely.

The following example illustrates the manner in which the above pairs of inverse networks may be used as fundamental rules for deriving the inverse circuits of more complex networks. Let it be required to find the inverse network corresponding to circuit A of Figure 163.

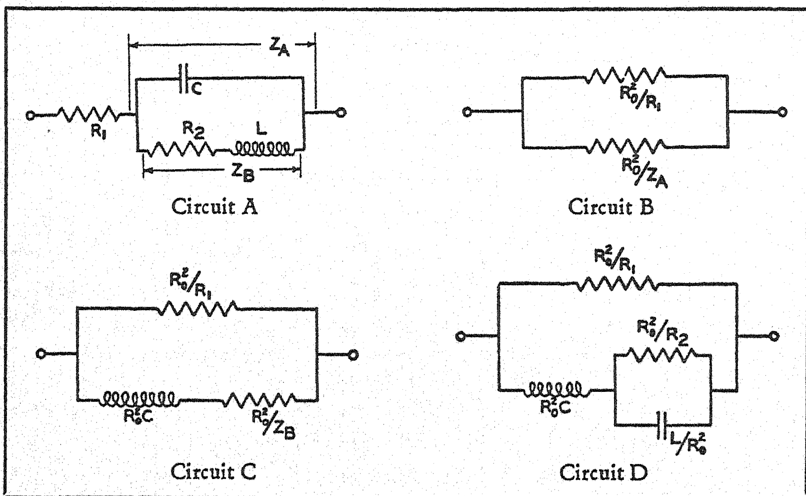
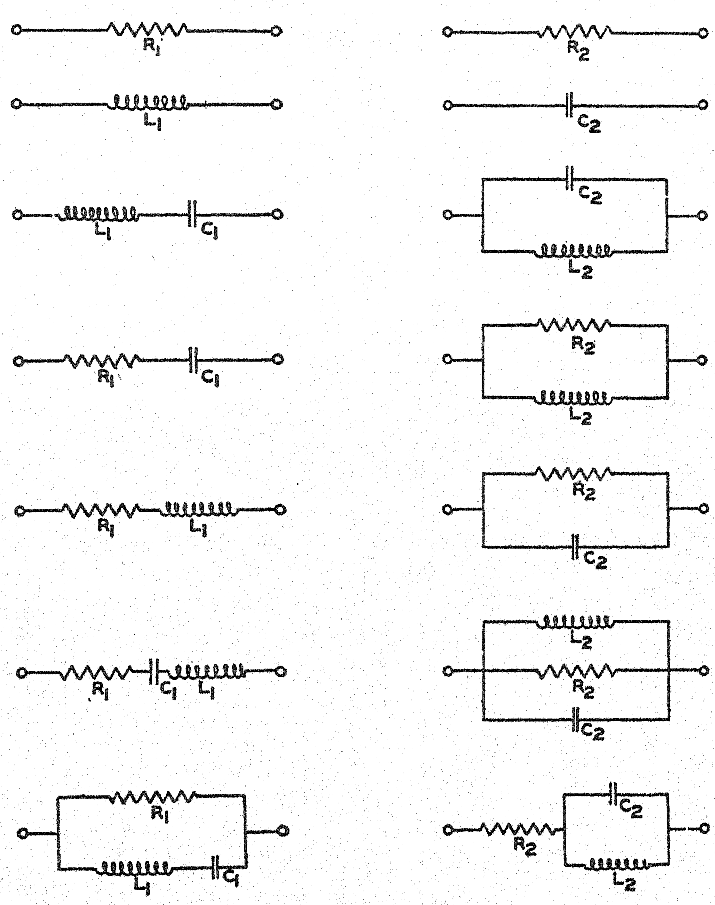


Figure 163.

Referring to Figure 163 it is seen that circuit A may be regarded as a resistance R_1 in series with an impedance Z_A. From Rule (3) the inverse of these two impedances in series is the parallel arrangement of B, where one of the impedances is the inverse of R_1, that is, R_0^2/R_1, and the other impedance is the inverse of Z_A which is R_0^2/Z_A. Now the inverse impedance R_0^2/Z_A may be simplified by regarding Z_A of circuit A as consisting of the condenser C in parallel with the impedance Z_B. Again making use of rule (3) the inverse of the impedance Z_A becomes an inductance of value $R^2{}_0C$ henries in series with the inverse of impedance Z_B which is R_0^2/Z_B. This condition is shown in circuit C. Once more a further simplification is obtained by using Rule (3) to find the inverse of Z_B which gives a resistance R_0^2/R_2 in parallel with a capacitance of L/R_0^2 farads. We are now able to write down circuit D

which is the final form for the inverse circuit corresponding to circuit A.

It is seen that the process of deriving an inverse network consists in replacing parallel circuits with series circuits and series circuits with parallel circuits and at the same time replacing all elements of the known network with their inverse elements in the inverse network. Experience in doing this makes it unnecessary to pass through the intermediate circuits as was done in the above. Figure 164 shows a number of pairs of inverse networks which will be useful for reference purposes in the equalizer work which follows.



$$R_1 R_2 = \frac{L_1}{C_2} = \frac{L_2}{C_1} = R_0^2.$$

Figure 164 — Inverse networks.

5. INSERTION LOSSES

The insertion loss of a network is a measure of its attenuation performance when operating under service conditions in a transmission facility. As already mentioned, networks usually operate between equal resistive impedances. For this condition the insertion loss is determined by the ratio of power delivered to the load before and after the network is inserted. Thus, if P_L is the power delivered to a load R_L before inserting the network and P'_L is the power after insertion of the network the insertion loss is defined as,

$$\text{I.L.} = 10 \log \frac{P_L}{P'_L} \qquad (68)$$

In cases where the load resistance R_L is not equal to the sending-end impedance R_G, the power P_L is usually defined as that delivered to the load through an ideal transformer which matches the two system impedances R_G and R_L. In other words, P_L is the maximum power which the source of supply can deliver to the load. Where R_G and R_L are equal the ideal matching transformer is not required. Insertion losses are defined in this manner in order that they will be losses for the insertion

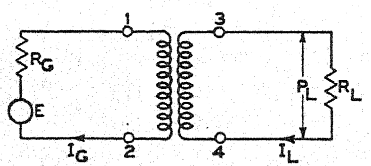


Figure 165-A — Ideal transformer matching R_G and R_L

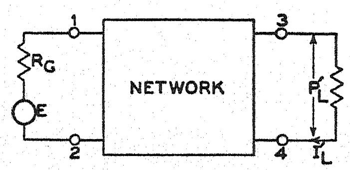


Figure 165-B.

of any network. That is, if P_L were not defined as the maximum power which could be delivered to the load, it would be possible to insert a network which under certain conditions would give an insertion gain. For some purposes this might not be objectionable but for the following wave filter and equalizer work it is desirable to exclude the possibility of an insertion loss becoming a gain.

Figure 165-A shows a cross-section of a transmission system having a sending impedance R_G and a receiving or load impedance R_L. The ideal transformer shown matches R_G with R_L so as to deliver the maximum power to R_L. For this reference circuit we have the following relations where P_G is the power sent into the transformer:

$$I_G = \frac{E}{2\,R_G}$$

$$P_G = \frac{E^2}{4\,R_G}$$

Now since the transformer is ideal it has zero dissipation loss and therefore all power sent into it is transferred on to the load. This gives

$$P_G = P_L = \frac{E^2}{4\,R_G} = I_L{}^2\,R_L$$

$$I_L = \sqrt{\frac{P_L}{R_L}} = \frac{E}{2\,\sqrt{R_L\,R_G}}$$

These are the reference power and reference current used in the computation of insertion losses. For the usual case where R_G and R_L are both equal resistances of some value, say R_0, the above equations reduce to the simple forms which would have been obtained had the ideal transformer not been used.

The power P'_L and the current I'_L refer to the condition where a network is inserted in the place of the ideal transformer, as illustrated in Figure 165-B. On a current basis the insertion loss formula corresponding to equation (68) becomes

$$\text{I.L.*} = 20 \log \frac{I_L}{I'_L} \tag{69}$$

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* NOTE: It will be noticed that there are two different notations used in this book to express insertion gains or insertion losses. Obviously, an insertion loss is a negative insertion gain, as an insertion gain is a negative insertion loss. For this reason, when using the decibel unit the loss or gain is expressed as a log function of either I'_L/I_L or I_L/I'_L. If the current I'_L (the load current after the change has been made in the circuit), is of a smaller numerical value than I_L, then an insertion loss has taken place. If the ratio I'_L/I_L had been used, a negative answer would have resulted, while if the ratio I_L/I'_L had been used, a positive answer would have resulted. These two answers would have the same numerical value but with opposite sign. In other words, either method is applicable as long as it is known whether the result is a gain or a loss.

In equalizer work where an insertion loss is always present, the ratio I_L/I'_L is used, as a positive answer results.

Chapter XVI

ATTENUATION EQUALIZERS

By HARRY KIMBALL

1. NATURE OF EQUALIZERS

As already mentioned, an attenuation equalizer is a four-terminal network whose attenuation loss, over a given frequency range, varies with frequency in some desirable manner. This means that if a number of frequencies of given amplitudes are simultaneously impressed upon the input terminals of an equalizer, the relative amplitudes of these frequencies will be changed when delivered to a load connected to the output terminals. The manner in which this change takes place is determined and can be controlled by the design of the equalizer. In sound picture work, the frequency range required extends from about 40 or 50 up to 7,500 or 8,000 cycles per second. This is called the transmission band. Any change in the relative amplitudes of the important frequency components of a signal changes the character of the reproduced sound as heard by the ear. For instance, where the upper frequencies of the transmission band are discriminated against, the signal is said to be "dull," and where they are accentuated, the signal is said to be "bright." In practice, the configurations required for the insertion loss curves of equalizers appear to vary over a wide range. Actually, many equalizer problems are but duplications of others with different values assigned to the network constants.

Equalizers may be provided with controls for varying at will the form of their insertion loss characteristics, or may be of fixed non-variable construction. Variable equalizers are used largely in re-recording work, whereas fixed equalizers are used as permanent parts of recording and reproducing equipments. The need for variable equalizers arises from the fact that in recording sound from artificial surroundings, it is, in some cases, not practicable to provide sets and pick-up conditions having the acoustical characteristics one would expect from the real scenes they represent when displayed as part of a continuous picture. The use of variable equalizers in the re-recording of a picture makes it possible to alter the amplitude relations of the frequency components of signals in a manner to create the illusion desired for the

picture, where, of course, this can be accomplished by such amplitude changes.

Except for the equalization employed in re-recording, sound systems are usually arranged to preserve, insofar as possible, the amplitude relations of the frequency components of signals. When unavoidable amplitude distortion occurs in one part of a system, fixed attenuation equalizers provide the means for making a permanent correction. In some cases attenuation distortion is deliberately inserted in one part of a system and compensated for in another part, for benefits obtained by removing the load from certain of the recording equipments as, for instance, for the the complementary method of recording discussed in Chapter III.

2. GENERAL EQUALIZER TYPES

From the great amount of work which has been done on the design theory of attenuation equalizers a number of general circuit arrangements have emerged which have proven to be the most satisfactory for general use. The network engineer does not necessarily restrict himself to the use of these few types but they do represent a large part of his kit of tools. These equalizer circuits are designated in the following manner:

(1) Series Impedance Type.
(2) Shunt Impedance Type.
(3) Full Series Type.
(4) Full Shunt Type.
(5) T Type.
(6) Bridged T Type.
(7) Lattice Type.

In general, the transmission characteristics of the circuit arrangements designated in the above manner are made to depend upon two impedances, denoted as Z_1 and Z_2, where Z_1 is usually a series impedance and Z_2 is a shunt impedance. Z_1 and Z_2 are defined as being the impedances of any pair of two-terminal networks which are inverse to each other with respect to the line impedance R_0; that is,

$$Z_1 Z_2 = R_0^2$$

In any given equalizer problem, Z_1 and Z_2 must necessarily be assigned specific circuit arrangements. The general properties of the above typical equalizer sections, however, can be determined before considering specific arrangements for Z_1 and Z_2. This is done in the work immediately following.

For each equalizer type we are principally interested in two items:

First, its insertion loss, and second, the impedance match conditions at the input and output terminals. These are determined for the above networks operating between circuit impedances of R_0 ohms at both the input and output terminals.

(a) Series Impedance Type

One of the simplest methods available for controlling, with respect to frequency, the power delivered to a load resistance from a resistive power source is to insert an impedance in series with one side of the connecting circuit in the manner shown in Figure 166-A. For this circuit the symbol, $2Z_1$, is used to designate the inserted impedance; that is, the equalizer consists of an impedance, $2\,Z_1$, connected between the 1-3 terminals and with the 2-4 terminals shorted together. Without the use of the equalizer the current which the power source can deliver to the load is

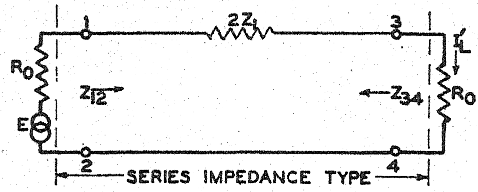


Figure 166-A — Series impedance type equalizer.

$$I_L = \frac{E}{2\,R_0}$$

Insertion of the equalizer changes this current to a value I'_L which is

$$I'_L = \frac{E}{2\,R_0 + 2\,Z_1}$$

The insertion loss by means of equation (69) then becomes

$$\text{I.L.} = 20 \log \frac{I_L}{I'_L} = 20 \log \frac{R_0 + Z_1}{R_0} \qquad (70)$$

Since the purpose of an equalizer is to secure an insertion loss which varies with frequency in some desirable manner, the impedance Z_1 must be a function of frequency to secure this effect. In other words, Z_1 cannot be entirely resistive because, if it were, the insertion loss would be constant with frequency.

The impedances looking into the network at the input and the output terminals are given by the expression

$$Z_{12} = Z_{34} = R_0 + 2\,Z_1 \qquad (71)$$

Obviously, this network does not match impedances at either the input or output terminals, and the mis-match obtained varies with frequency since Z_1 is a frequency function.

(b) Shunt Impedance Type

Another four-terminal equalizer circuit quite similar to the previous case is obtained by the use of a shunt impedance in the manner shown in Figure 166-B. For this circuit the symbol $Z_2/2$ is used to designate the shunt impedance. The load current I_L received when the equalizer is not in the circuit is again equal to $(E/2\ R_0)$, and the load current I'_L obtained when the equalizer is inserted is

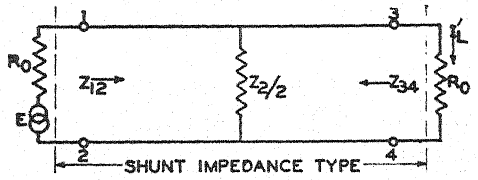


Figure 166-B — Shunt impedance type equalizer.

$$I'_L = \frac{E}{R_0 + \dfrac{R_0 Z_2}{2\ R_0 + Z_2}} \frac{Z_2}{2\ R_0 + Z_2} = \frac{E}{2\ R_0} \frac{Z_2}{(R_0 + Z_2)}$$

The insertion loss for the network then becomes

$$\text{I.L.} = 20 \log \frac{I_L}{I'_L} = 20 \log \frac{R_0 + Z_2}{Z_2} \tag{72}$$

Since the impedance Z_1 used in connection with the series impedance type circuit has been defined as being the inverse of Z_2, that is, $Z_1 Z_2 = R_0^2$, we may also write for equation (72)

$$\text{I.L.} = 20 \log \frac{R_0 + Z_1}{R_0} \tag{73}$$

This means that the series impedance and shunt impedance type equalizers have the same insertion loss characteristics when designed with inverse impedances as their variable elements.

The impedance looking into the network either at the input terminals with the output terminals connected to R_0 ohms, or at the output terminals with the input terminals connected to R_0 ohms is,

$$Z_{12} = Z_{34} = \frac{\dfrac{R_0 Z_2}{2}}{R_0 + \dfrac{Z_2}{2}} = \frac{R_0 Z_2}{2\ R_0 + Z_2} \tag{74}$$

This network does not provide a match of impedances at the input and output terminals as can be seen, of course, from inspection.

(c) Full Series Constant Resistance Type

This equalizer circuit shown in Figure 166-C is quite commonly used in practice. As Figure 166-C shows, both the inverse impedances, Z_1 and Z_2, are used in this circuit. Referring to the figure, the impedance looking in at the 1-2 terminals with the 3-4 terminals connected to R_0 resistive ohms, is

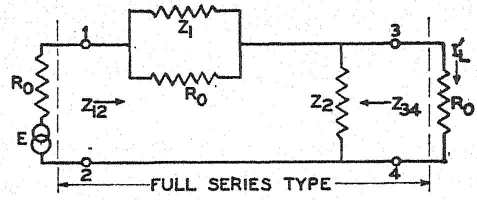


Figure 166-C — Full series constant resistance type equalizer.

$$Z_{12} = \frac{R_0 Z_1}{R_0 + Z_1} + \frac{R_0 Z_2}{R_0 + Z_2} = R_0 \frac{R_0 (Z_1 + Z_2) + 2 Z_1 Z_2}{R_0 (Z_1 + Z_2) + R_0^2 + Z_1 Z_2}$$

But by definition, $Z_1 Z_2 = R_0^2$ and we therefore have

$$Z_{12} = R_0 \tag{75}$$

The impedance Z_{34} looking into the network at the 3-4 terminals with the 1-2 terminals connected to a resistance of R_0 ohms is

$$Z_{34} = \frac{Z_2 \left[R_0 + \dfrac{R_0 Z_1}{R_0 + Z_1} \right]}{Z_2 + R_0 + \dfrac{R_0 Z_1}{R_0 + Z_1}}$$

$$Z_{34} = \frac{R_0 Z_2 (R_0 + 2 Z_1)}{(R_0 + Z_2)\ (R_0 + Z_1) + R_0 Z_1} \tag{76}$$

This impedance formula can be put in various forms by different transformations, but since it is of little importance these are omitted here. The important item in connection with the terminal impedances is that Z_{12} is a constant resistance regardless of frequency, whereas Z_{34} varies with frequency in some manner. This network then provides perfect impedance conditions at the 1-2 terminals, but not at the 3-4 terminals. The making of Z_1 and Z_2 inverse to each other makes it possible to secure this resistive impedance at the 1-2 terminals.

With regard to the insertion loss of the network the current delivered to the load resistance by the network is:

$$I'_L = \frac{E}{2 R_0} \frac{Z_2}{R_0 + Z_2}$$

Since the current without the use of the network is $I_L = \frac{E}{2\,R_0}$ the insertion loss becomes

$$\text{I.L.} = 20 \log \frac{R_0 + Z_2}{Z_2} = 20 \log \frac{R_0 + Z_1}{R_0} \qquad (77)$$

(d) Full Shunt Constant Resistance Type

This network shown in Figure 166-D is the inverse of the full series type network with respect to the 1-2 terminals. The terminal impedance and insertion loss formulae are derived in the same manner. It will be seen that in this case also a match of impedances is obtained at the 1-2 terminals but not at the 3-4 terminals.

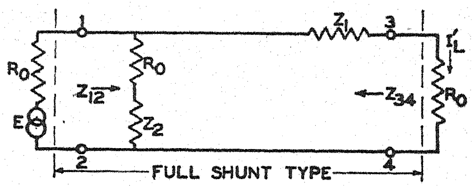


Figure 166-D — Full shunt constant resistance type equalizer.

$$Z_{12} = \frac{(R_0 + Z_2)\ (R_0 + Z_1)}{2\,R_0 + Z_1 + Z_2} = \frac{R_0^2 + Z_1 Z_2 + R_0\,(Z_1 + Z_2)}{2\,R_0 + Z_1 + Z_2}$$

But

$$Z_1\,Z_2 = R_0^2$$

$$Z_{12} = R_0 = \text{constant resistance} \qquad (78)$$

$$Z_{34} = \text{not constant with frequency}$$

$$I'_L = \frac{E}{2\,R_0}\,\frac{R_0 + Z_2}{2\,R_0 + Z_1 + Z_2} = \frac{E}{2\,R_0}\,\frac{R_0 + Z_2}{2\,R_0 + Z_2 + \frac{R_0^2}{Z_2}}$$

$$= \frac{E}{2\,R_0}\,\frac{Z_2\,(R_0 + Z_2)}{R_0^2 + 2\,R_0\,Z_2 + Z_2^2} = \frac{E}{2\,R_0}\,\frac{Z_2\,(R_0 + Z_2)}{(R_0 + Z_2)^2}$$

$$I'_L = \frac{E}{2\,R_0}\,\frac{Z_2}{R_0 + Z_2}$$

$$I_L = \frac{E}{2\,R_0}$$

Insertion Loss =

$$20 \log \frac{I_L}{I'_L} = 20 \log \frac{R_0 + Z_1}{R_0} = 20 \log \frac{R_0 + Z_2}{Z_2} \qquad (79)$$

(e) T Type Constant Resistance Equalizer

This T type network of Figure 166-E may be analyzed by separating it into two networks in tandem, one a full series type and the other a

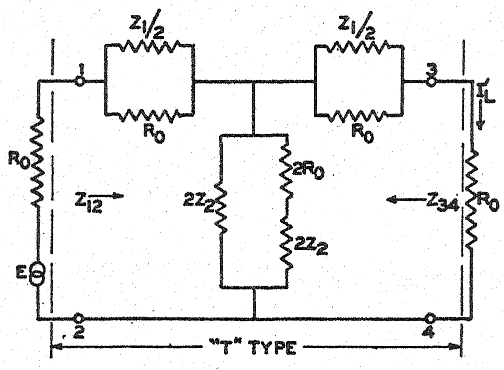


Figure 166-E — T type constant resistance type equalizer.

full shunt type as shown in Figure 166-F.

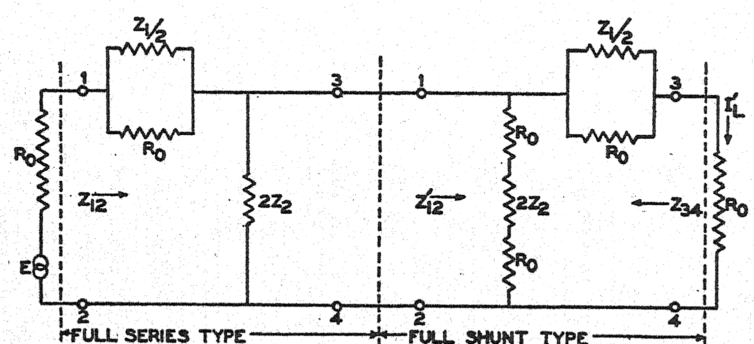


Figure 166-F — Equivalent circuit for T type equalizer of Figure 166-E.

For the full shunt type:

$$Z'_{12} = R_0 \qquad \text{[using formula (78)]}$$

$$\text{I.L.} = 20 \log \frac{2\,R_0 + 2\,Z_2}{R_0 + 2\,Z_2} \qquad \text{[using formula (79)]}$$

For the full series type:

$$Z_{12} = R_0 \qquad \text{[using formula (75)]}$$

$$\text{I.L.} = 20 \log \frac{R_0 + 2\,Z_2}{2\,Z_2} \qquad \text{[using formula (77)]}$$

From this data and because the T network is symmetrical, we have for the T network of Figure 166-E

$$Z_{12} = Z_{34} = R_0 = \text{constant resistance at both ends} \qquad (80)$$

And the insertion loss is the sum of the losses for the two component sections; that is,

$$\text{I.L.} = 20 \log \frac{2\,R_0 + 2\,Z_2}{R_0 + 2\,Z_2} + 20 \log \frac{R_0 + 2\,Z_2}{2\,Z_2}$$

$$\text{I.L.} = 20 \log \left(\frac{2\,R_0 + 2\,Z_2}{R_0 + 2\,Z_2}\right)\left(\frac{R_0 + 2\,Z_2}{2\,Z_2}\right)$$

$$\text{I.L.} = 20 \log \frac{R_0 + Z_2}{Z_2} = 20 \log \frac{R_0 + Z_1}{R_0} \qquad (81)$$

(f) Bridged T Constant Resistance Equalizer

As a means of deriving the impedance and loss characteristics of the bridged T equalizer of Figure 166-G, consider again the full shunt type equalizer arranged as in Figure 166-H.

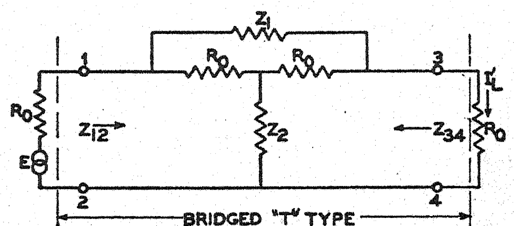


Figure 166-G — Bridged T constant resistance type equalizer.

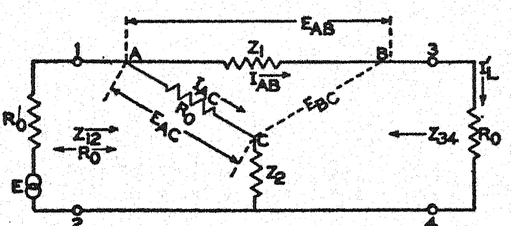


Figure 166-H — Equivalent circuit for bridged T type equalizer of Figure 166-G.

By Kirchoff's laws the voltage E_{BC} is:

$$E_{BC} = E_{AB} - E_{AC}$$

$$E_{BC} = Z_1 I_{AB} - R_0 I_{AC}$$

Since the impedance looking into the network at the 1-2 terminals is R_0, the voltage applied at these terminals is $E/2$. This gives for I_{AB} and I_{AC}:

$$I_{AB} = \frac{E}{2\ (R_0 + Z_1)}$$

$$I_{AC} = \frac{E}{2\ (R_0 + Z_2)}$$

Then the voltage E_{BC} becomes

$$E_{BC} = \frac{E}{2}\left[\frac{Z_1}{R_0 + Z_1} - \frac{R_0}{R_0 + Z_2}\right]$$

$$E_{BC} = \frac{E}{2}\left[\frac{Z_1 Z_2 - R_0^2}{(R_0 + Z_1)\ (R_0 + Z_2)}\right] \text{and as } Z_1 Z_2 = R_0^2$$

$$E_{BC} = \text{zero}$$

Since the voltage existing across the points B and C is zero, an impedance of any value may be connected between the points without disturbing the network. When a resistance of R_0 ohms is connected between the points B and C we arrive at the bridged T network of Figure 166-H. Hence, the insertion loss for the bridged T network is the same as for the full shunt circuit. Since the bridged T network is symmetrical at both ends, we know that $Z_{34} = Z_{12}$. Then we finally have for the bridged T network of Figure 166-G:

$$Z_{12} = Z_{34} = R_0 = \text{constant resistance} \tag{82}$$

and

$$\text{I.L.} = 20 \log \frac{R_0 + Z_1}{R_0} = 20 \log \frac{R_0 + Z_2}{Z_2} \tag{83}$$

(g) Bridge or Lattice Type Equalizer

Before taking up the lattice type equalizer, it will be useful to work out the current relations of the following bridge circuit when a voltage E is applied directly to one pair of the diagonal terminals and a resistance R_0 is connected across the other diagonal terminals. The impedances Z_A and Z_B shown in the figure are defined as being inverse to each other with respect to R_0; that is, $Z_A Z_B = R_0^2$. Since the configuration of the above circuit is symmetrical, it is evident that equal currents flow through each of the Z_A impedances, and similarly for the Z_B impedances. By Kirchoff's laws, the voltage drop around any closed loop is zero. Considering the closed loop 1-3-4-1,

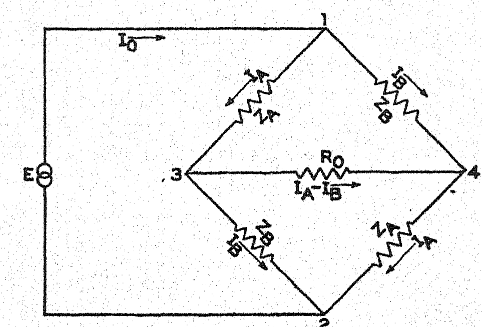


Figure 166-I — Bridge circuit.

we have,

$$I_A Z_A + (I_A - I_B) R_0 - I_B \frac{R_0^2}{Z_A} = 0$$

$$I_A (Z_A + R_0) - I_B R_0 \left(\frac{Z_A + R_0}{Z_A}\right) = 0$$

or we have

$$I_A Z_A = I_B R_0 \tag{84}$$

Now considering the loop 1-3-2-1, which includes the voltage E, we have

$$I_A Z_A + I_B \frac{R_0^2}{Z_A} = E \tag{85}$$

Substituting $I_B R_0$ for $I_A Z_A$ gives

$$I_B = \frac{E}{R_0} \frac{Z_A}{R_0 + Z_A}$$

Then from equation (84)

$$I_A = \frac{E}{R_0} \frac{R_0}{R_0 + Z_A}$$

$$I_0 = I_A + I_B = \frac{E}{R_0}\left[\frac{R_0}{R_0 + Z_A} + \frac{Z_A}{R_0 + Z_A}\right] = \frac{E}{R_0} \tag{86}$$

$$I_A - I_B = \frac{E}{R_0} \frac{R_0 - Z_A}{R_0 + Z_A} = I_0 \frac{R_0 - Z_A}{R_0 + Z_A} \tag{87}$$

It is noted from equation (86) that the impedance looking into the bridge at terminals 1-2 is a constant resistance of R_0 ohms. Also, from equation (87) the current delivered to the resistance R_0 is the current I_0 entering terminals 1-2 multiplied by the factor $\frac{R_0 - Z_A}{R_0 + Z_A}$

We are now in a position to consider the lattice type equalizer section of Figure 166-J. This network is the same as the above bridge circuit except that a resistance of R_0 ohms has been inserted in series with the voltage E, and the impedances Z_A and Z_B have been replaced by

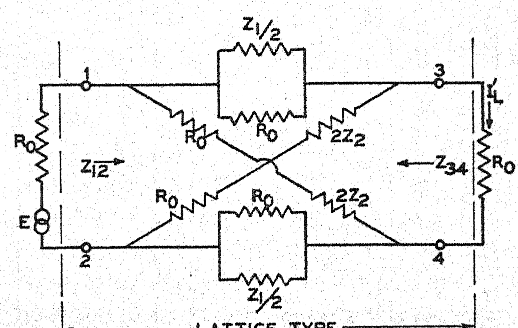
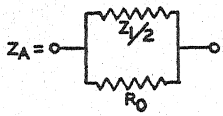
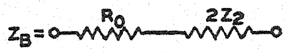


Figure 166-J — Lattice type equalizer.

the following circuits where Z_1 and Z_2 are inverse to each other:





Since the network is symmetrical about the 1-2 and 3-4 terminals we may say by means of equation (86):

$$Z_{12} = Z_{34} = R_0 \tag{88}$$

and also from equation (87)

$$I'_L = \frac{E}{2\,R_0}\,\frac{R_0 - Z_A}{R_0 + Z_A}$$

$$I'_L = \frac{E}{2\,R_0}\,\frac{R_0 - \dfrac{R_0\,Z_1}{2\,R_0 + Z_1}}{R_0 + \dfrac{R_0\,Z_1}{2\,R_0 + Z_1}} = \frac{E}{2\,(R_0 + Z_1)}$$

Since $I_L = \dfrac{E}{2\,R_0}$ we have for the insertion loss

$$\text{I.L.} = 20 \log \frac{R_0 + Z_1}{R_0} = 20 \log \frac{R_0 + Z_2}{Z_2} \tag{89}$$

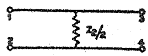
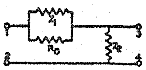
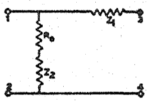
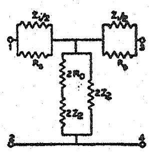
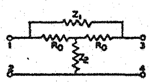
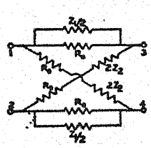
3. SUMMARY OF EQUALIZER TYPES

Figure 166 is a summary of the seven equalizer types discussed above. It is noted that the insertion loss formula, as expressed by the equation

$$\text{I.L.} = 20 \log \frac{R_0 + Z_1}{R_0} = 20 \log \frac{R_0 + Z_2}{Z_2} \tag{90}$$

is applicable to each of the equalizer types. This means that an insertion loss characteristic obtained with one of the equalizer types can be duplicated by any of the other types. The above formula also shows that the configurations of the insertion loss characteristics obtained for the equalizers of Figure 166 are determined solely by the inverse arms of the networks as represented by impedances Z_1 and Z_2. This feature makes it practicable, in a design problem, to determine the circuits of the inverse arms independent of the equalizer types with which they are to be used. In the work immediately following, the insertion loss

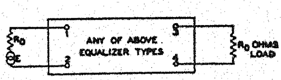
Figure 166 — Fundamental Equalizer Types

Figure	Network	Type	Z_{12}	Z_{34}	Insertion Loss
166-A		Series Imp.	Not Constant	Not Constant	$20 \log \frac{R_0 + Z_1}{R_0}$
166-B		Shunt Imp.	Not Constant	Not Constant	$20 \log \frac{R_0 + Z_2}{Z_2}$
166-C		Full Series	R_0	Not Constant	$20 \log \frac{R_0 + Z_1}{R_0}$
166-D		Full Shunt	R_0	Not Constant	$20 \log \frac{R_0 + Z_1}{R_0}$
166-E		T Type	R_0	R_0	$20 \log \frac{R_0 + Z_1}{R_0}$
166-G		Bridged T	R_0	R_0	$20 \log \frac{R_0 + Z_1}{R_0}$
166-J		Lattice	R_0	R_0	$20 \log \frac{R_0 + Z_1}{R_0}$

NOTES:

(1) $Z_1 Z_2 = R_0^2$ for all networks

(2) $20 \log \frac{R_0 + Z_1}{R_0} = 20 \log \frac{R_0 + Z_2}{Z_2}$

(3) Working Circuit = 

characteristics obtained with the use of certain common circuit arrangements for the inverse arms are discussed in detail. In doing this it is convenient to consider two types of circuits; first, those containing only reactive elements, and second, those containing both resistive and reactive elements. The purpose of this classification is made clear as we progress.

As already mentioned, where the inverse arms are entirely resistive, the seven equalizer types degenerate into attenuators. For instance, suppose we let

$$Z_1 = R_0\,(K-1) \text{ and } Z_2 = \frac{R_0}{K-1} \text{ (ohms)}$$

where K is a constant to be defined later. Use of these arms with the seven circuits of Figure 166 results in the attenuator circuits of Figure 167. The insertion loss of these attenuators is

$$\text{I.L.} = 20 \log \frac{R_0 + R_0\,(K-1)}{R_0} = 20 \log K$$

and K may now be seen to be the factor which determines the loss of the attenuators.

Another important feature in connection with the above equalizer circuits is the impedance match obtained at the 1-2 and 3-4 terminals. It is seen that two of the types shown do not give a match at either end; two produce a match of impedances at one end; and the remaining three

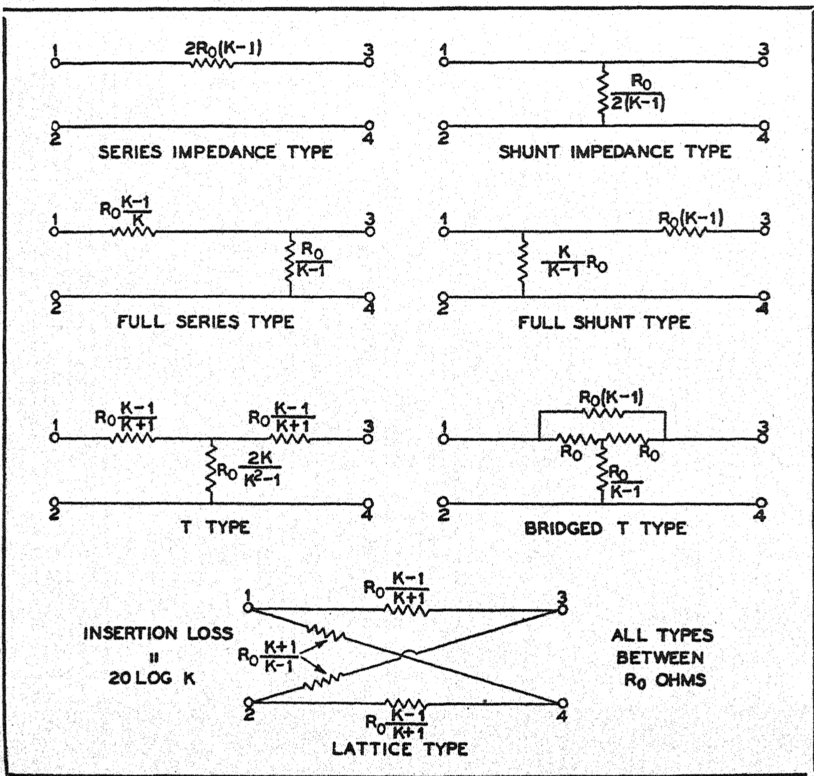


Figure 167 — Attenuator circuits.

provide a match of impedances at both ends. This dissimilarity between the networks makes one circuit more desirable than another under different conditions as discussed later.

4. INVERSE ARMS PURELY REACTIVE

When the inverse arms of equalizers consist only of reactive elements their impedances may be expressed as

$$Z_1 = jX_1 \text{ and } Z_2 = jX_2$$

where $(jX_1)\ (jX_2) = R_0^2$. Substituting these values into the general loss formula of equation (90) we have

$$\text{I.L.} = 20 \log \frac{R_0 + jX_1}{R_0}$$

The term $\left(\frac{R_0 + jX_1}{R_0}\right)$ represents the vector ratio of two currents, I_L and I'_L. A network insertion loss is proportional to the ratio of the absolute magnitudes of the two currents, which is expressed by the absolute magnitude of the vector $\left(\frac{R_0 + jX_1}{R_0}\right)$

This gives:

$$\text{I.L.} = 20 \log \sqrt{\frac{R_0^2 + X_1^2}{R_0^2}}$$

$$\text{I.L.} = 10 \log \frac{R_0^2 + X_1^2}{R_0^2} = 10 \log \frac{R_0^2 + X_2^2}{X_2^2} \tag{91}$$

Since X_1 and X_2 are pure reactances they vary with frequency between values of zero and infinity. This causes the insertion loss of equalizers employing such arms to also vary between zero and infinity over the complete frequency spectrum. It is useful to note at this time that when X_1 has a numerical value of R_0 ohms, and X_2 is also equal to R_0 ohms, the insertion loss becomes

$$\text{I.L.} = 10 \log \frac{R_0^2 + R_0^2}{R_0^2} = 10 \log 2 = 3 \text{ db} \tag{92}$$

In other words, the insertion loss passes through the three db loss point whenever the reactances X_1 and X_2 are numerically equal to R_0 ohms. This provides a useful design relation which will be used later.

Formula (91) is, of course, valid for inverse arms of any complexity. In practice, a few relatively simple pairs are sufficient for

handling a large portion of the equalizer problems encountered. In the following work, insertion loss data are given for the pairs of reactive

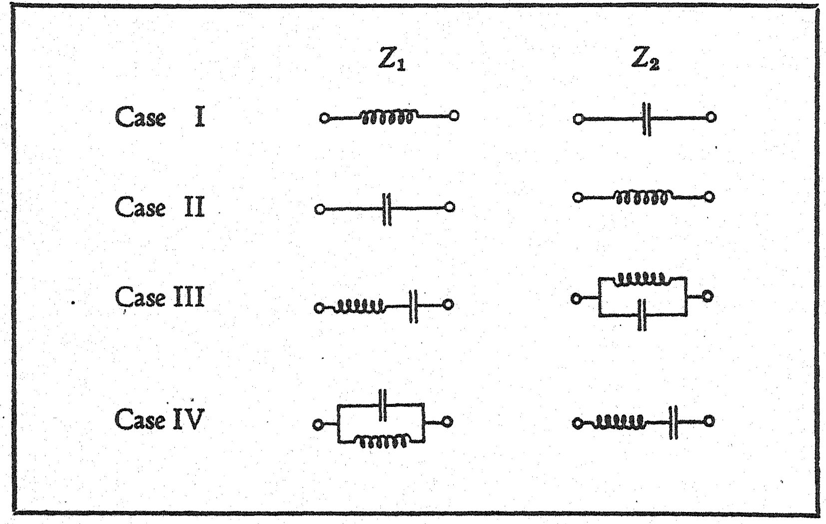
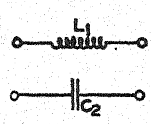


Figure 168 — Pairs of reactive inverse networks.

inverse networks of Figure 168, used in connection with any of the equalizer types of Figure 166.

(a) Case I

When $Z_1 =$ and $Z_2 =$



From inspection it can be seen that the insertion loss of equalizers employing inverse arms of this type increases with increasing frequency. For example, Figure 169 shows the full series type equalizer of Figure 166-C employing these inverse arms. Since the impedance of the shunt

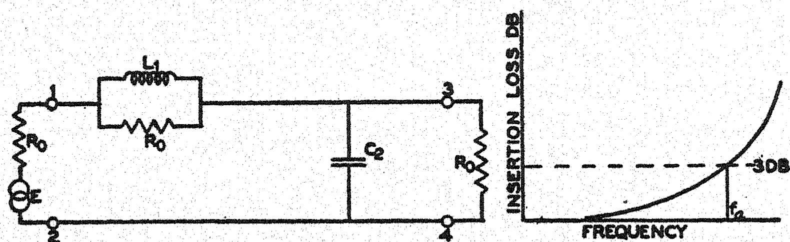


Figure 169 — A full series, Case I, equalizer with insertion loss characteristic.

condenser C_2 decreases with frequency, and that of the series inductance L_1 increases, the insertion loss becomes greater as the frequency increases.

This effect can be shown analytically in a more precise manner as follows:

From equation (48) the reactance of the inductance L_1 can be written

$$jX_1 = jX_S \frac{f}{f_S} \tag{93}$$

where X_S is the reactance at the frequency f_S. It was shown in equation (92) that where equalizers employ purely reactive inverse arms their insertion loss is three db at frequencies which makes the reactances of the inverse arms each numerically equal to R_0 ohms. Let "f_a" be the frequency where the reactances X_1 and X_2 are each R_0 ohms, or, in other words, f_a is the frequency where equalizers employing these inverse arms have three db loss. We have

$$jR_0 = jX_S \frac{f_a}{f_S} \text{ or } \frac{X_S}{f_S} = \frac{R_0}{f_a}$$

Then equation (93) can be written

$$jX_1 = jR_0 \frac{f}{f_a}$$

Using this value of X_1 in connection with equation (91) gives for the final insertion loss formula

$$\text{I.L.} = 10 \log \left[1 + \left(\frac{f}{f_a} \right)^2 \right] \tag{94}$$

Chart II shows this insertion loss formula graphically where loss in db is plotted against the frequency ratio f/f_a. As would be expected when $f/f_a = 1$ or $f = f_a$, the insertion loss is three db and the amount of loss increases with increasing frequency. In using this chart in practical problems we need to specify only the frequency f_a where we wish three db loss to occur, and read the losses for other frequencies from the chart after determining the corresponding frequency ratios.

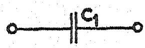
In terms of R_0 and the design constant f_a, the values of L_1 and C_1 can be obtained by the use of equations (47) and (50). We have

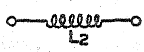
$$\begin{aligned} L_1 &= \frac{X_S}{\omega_S} = \frac{R_0}{\omega_a} = \frac{R_0}{2\pi f_a} \\ C_2 &= \frac{1}{X_S\, \omega_S} = \frac{1}{R_0\, \omega_a} = \frac{1}{2\pi f_a R_0} \end{aligned} \tag{95}$$

Column I of Chart I shows the equalizer sections obtained when this pair of inverse arms is used with the seven fundamental equalizer

types. The formulae for computing the electrical constants for each of the networks are given at the bottom of the column.

(b) Case II

When $Z_1 =$ 

and $Z_2 =$ 

This pair of inverse arms is the same as the previous pair except that Z_1 and Z_2 are interchanged. Figure 170 shows a full series type equalizer

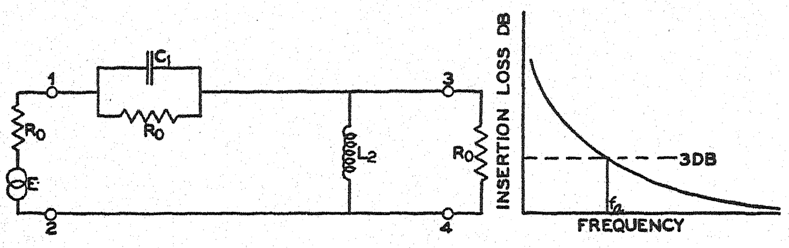


Figure 170 — A full series, Case II, equalizer with insertion loss characteristic.

employing these arms. From inspection it can be seen that the insertion loss of this equalizer should decrease with increasing frequency somewhat as indicated in the figure. The insertion loss formula expressing this characteristic is obtained in the following manner:

The reactance of the inductance L_2 is

$$jX_2 = jR_0 \frac{f}{f_a}$$

where f_a is the frequency for which the numerical value of X_2 is R_0 ohms. Substituting this value of X_2 into equation (91) we have for the insertion loss formula of equalizers employing these arms:

$$\text{I. L.} = 10 \log \left[\frac{R_0^2 + R_0^2 \left(\frac{f}{f_a}\right)^2}{R_0^2 \left(\frac{f}{f_a}\right)^2} \right]$$

$$\text{I. L.} = 10 \log \left[1 + \left(\frac{f_a}{f}\right)^2 \right] \qquad (96)$$

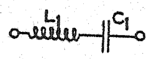
and the values of L_2 and C_1 in terms of R_0 and f_a as obtained from equations (48) and (50) are

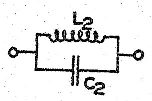
$$L_2 = \frac{X_S}{\omega_S} = \frac{R_0}{\omega_a} = \frac{R_0}{2\,\pi\, f_a}$$

$$C_1 = \frac{1}{\omega_S\, X_S} = \frac{1}{R_0\, \omega_a} = \frac{1}{2\,\pi\, f_a\, R_0} \qquad (97)$$

Chart III shows the insertion loss of equation (96) plotted against the ratio f/f_a. It is seen that in this case the insertion loss decreases with increasing frequency. Again, as in the previous case, the loss when $f = f_a$ is three db. Column II of Chart I shows the equalizer circuits obtained when this pair of inverse arms is used with the fundamental types of Figure 166, and the electrical constants for each of the circuits are given at the bottom of the column. In practical work, the frequency f_a is taken as the point where we wish three db loss to occur, and the losses at other frequencies are read from the curve after computing the corresponding frequency ratios.

(c) Case III

When $Z_1 =$ 

and $Z_2 =$ 

The general form of the insertion loss characteristic of equalizers using this pair of inverse arms is shown in Figure 171. In this figure, the circuit given is obtained by using the above inverse arms with the full

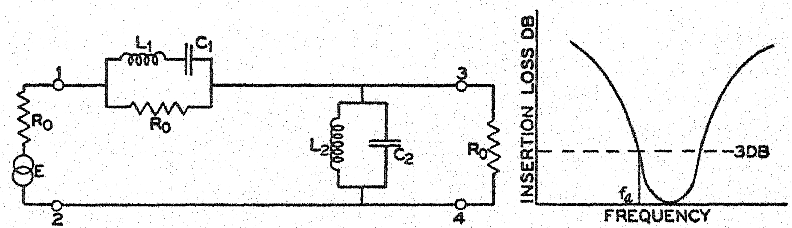


Figure 171 — A full series, Case III, equalizer with insertion loss characteristic.

series type equalizer of Figure 166-C. Since Z_1 and Z_2 are inverse to each other, L_1 and C_1 are resonant at the same frequency, f_R, that L_2 and C_2 are anti-resonant. When this occurs, the network insertion loss is zero as shown in the figure. For frequencies less or greater than f_R, the loss is finite and increases with frequency departure from f_R. At some frequency f_a, which is the frequency for which Z_1 and Z_2 are each numerically equal to R_0 ohms, the network loss is three db. The formula for precisely expressing this insertion loss characteristic is obtained as follows:

Letting $f_S = f_a$ and $X_S = R_0$ as was done in the foregoing cases, we have by means of equation (55)

$$jX_1 = jR_0 \frac{\dfrac{f}{f_R} - \dfrac{f_R}{f}}{a - \dfrac{1}{a}} \tag{98}$$

where $s = a$, and "a" is the frequency ratio f_R/f_a and by definition is greater than unity. Substitution of this value of X_1 into equation (91) gives for the insertion loss equation:

$$\text{I. L.} = 10 \log \left[1 + \frac{\left(\frac{f}{f_R} - \frac{f_R}{f} \right)^2}{\left(a - \frac{1}{a} \right)^2} \right] \tag{99}$$

This equation expresses the insertion loss of equalizers using these arms in terms of the resonant frequency f_R and the frequency f_a, where three db loss is obtained. The electrical elements, L_1, C_1, L_2 and C_2, do not appear in this insertion loss formula, but may also be expressed in terms of these design parameters. Remembering that $X_S = R_0$, $\omega_S = \omega_a$ and

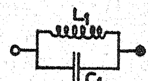
$\frac{L_1}{C_2} = \frac{L_2}{C_1} = R_0^2$ we obtain from equation (56) and (57)

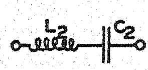
$$L_1 = \frac{R_0}{\omega_a} \frac{1}{a^2 - 1} \qquad L_2 = \frac{R_0}{\omega_a} \frac{a^2 - 1}{a^2}$$

$$C_1 = \frac{1}{\omega_a R_0} \frac{a^2 - 1}{a^2} \qquad C_2 = \frac{1}{\omega_a R_0} \frac{1}{a^2 - 1} \tag{100}$$

Chart IV shows graphs of the insertion loss formula of equation (99) for a number of arbitrarily chosen values of "a". Assigning a specific value to "a" is equivalent to specifying the relation of the resonant frequency f_R to the frequency f_a where three db loss is obtained. For instance, for the curve where $a = 2$, f_a is one-half the frequency of resonance. Column III of Chart I shows the equalizers obtained when this pair of inverse arms is used with each of the network types of Figure 166. The formulae for the electrical constants of each of the circuits appear at the bottom of the column.

(d) Case IV

When $Z_1 =$ 

and $Z_2 =$ 

This pair of inverse arms is the same as the previous pair except that Z_1 and Z_2 are interchanged. Figure 172 shows a full series type equalizer employing these arms, and the general form of the associated insertion loss characteristic. It is seen that the loss is infinite at the resonant frequency f_R, and is zero at zero frequency and at an infinite frequency. At some frequency f_a, where both Z_1 and Z_2 are equal to

R_0 ohms, the loss curve passes through the three db point. As in the previous cases, f_a can be placed at any point in the frequency range be-

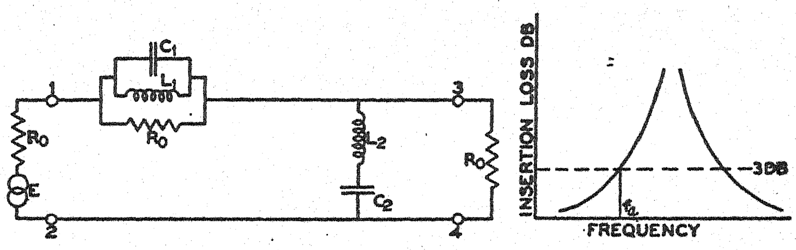


Figure 172 — A full series, Case IV, equalizer with insertion loss characteristic.

low f_R, simply by arranging Z_1 and Z_2 to be numerically equal to R_0 ohms at this frequency.

The insertion loss formula is obtained as follows: Letting $f_S = f_a$, $X_S = R_0$, and $s = a$, we have by means of equation (55)

$$jX_2 = jR_0 \frac{\dfrac{f}{f_R} - \dfrac{f_R}{f}}{a - \dfrac{1}{a}}$$

Substituting this value of X_2 into equation (91) gives

$$\text{I.L.} = 10 \log \left[1 + \frac{\left(a - \dfrac{1}{a}\right)^2}{\left(\dfrac{f}{f_R} - \dfrac{f_R}{f}\right)^2} \right] \tag{101}$$

By means of equations (56) and (57), and remembering that X_S and ω_S have been assigned the values of $X_S = R_0$ and $\omega_S = \omega_a$, we have for L_2 and C_2

$$L_2 = \frac{R_0}{\omega_a} \frac{1}{a^2 - 1} \qquad C_2 = \frac{1}{R_0\,\omega_a} \frac{a^2 - 1}{a^2} \tag{102}$$

And by the inverse relations of $\dfrac{L_1}{C_2} = \dfrac{L_2}{C_1} = R_0^2$ we have

$$L_1 = \frac{R_0}{\omega_a} \frac{a^2 - 1}{a^2}$$

$$C_1 = \frac{1}{R_0\,\omega_a} \frac{1}{a^2 - 1} \tag{103}$$

Again, as in the previous cases, equation (101) expresses the insertion loss of equalizers using this pair of inverse arms in terms of f_R and f_a, that is, in terms of f/f_R and $f_R/f_a = a$. Chart V shows graphs of equation (101) for different values of a. Column IV of Chart I shows the equalizers obtained when this pair of inverse arms is used with each of

the equalizer types of Figure 166. Formulae for the electrical constants are given at the bottom of the column.

5. INVERSE ARMS CONTAINING BOTH RESISTIVE AND REACTIVE ELEMENTS

Where the inverse arms of equalizers contain both resistive and reactive elements the circuit configurations considered must again be limited to those most commonly used to keep the data presented within reason. The general circuit configurations of Figure 173 will first be discussed, after which specific circuits will be taken up. The significance of the symbol "K," which determines the value of the resistance, $R_0\ (K-1)$, will be made clear later. The first pair of inverse arms of Figure 173 is included only for the purposes of generality and as shown below contributes nothing new to the design of equalizers.

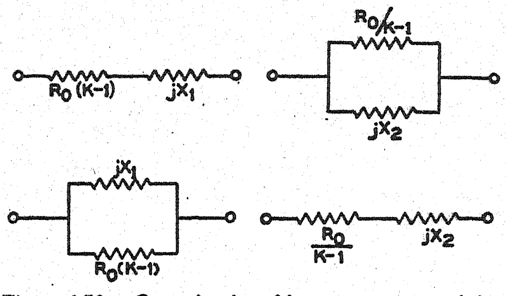


Figure 173—General pairs of inverse arms containing both resistive and reactive elements.

For this pair of arms we have

$$Z_1 = R_0\ (K-1) + jX_1$$

and when used in connection with equation (90) the insertion loss formula becomes

$$\text{I.L.} = 20 \log \frac{R_0 + R_0\ (K-1) + jX_1}{R_0}$$

$$= 20 \log \frac{R_0 K + jX_1}{R_0}$$

$$= 20 \log K \frac{R_0 + j\dfrac{X_1}{K}}{R_0}$$

$$= 20 \log K + 20 \log \frac{R_0 + j\dfrac{X_1}{K}}{R_0} \qquad (104)$$

This formula makes it clear that an equalizer using this pair of inverse arms is equivalent to a circuit consisting of an attenuator of loss equal to $20 \log K$, in tandem with an equalizer employing purely reactive inverse arms of values

$$Z_1 = j\frac{X_1}{K} \text{ and } Z_2 = jKX_2$$

This equivalent arrangement for a bridged T type equalizer is shown in Figure 174. Since the insertion loss performance of both of these

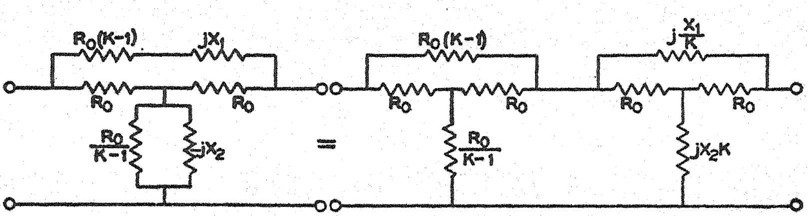


Figure 174 — Equivalent bridged T equalizer circuits.

component structures has already been discussed, equalizers employing the first pair of inverse arms of Figure 173 are not discussed further.

Referring to the second pair of inverse arms of Figure 173 we have

$$Z_1 = \frac{jR_0 X_1 (K-1)}{R_0 (K-1) + jX_1}$$

Used in connection with equation (90) this gives for the insertion loss

$$\text{I.L.} = 20 \log \frac{R_0 (K-1) + jKX_1}{R_0 (K-1) + jX_1}$$

Eliminating the vector symbol "j" by the method used in developing equation (91), this becomes

$$\text{I.L.} = 10 \log \frac{R_0^2 (K-1)^2 + K^2 X_1^2}{R_0^2 (K-1)^2 + X_1^2}$$

$$\text{I.L.} = 10 \log \left[1 + \frac{K^2 - 1}{1 + (K-1)^2 \left(\frac{R_0}{X_1}\right)^2} \right] \tag{105}$$

or expressed in terms of X_2 we have

$$\text{I.L.} = 10 \log \left[1 + \frac{K^2 - 1}{1 + (K-1)^2 \left(\frac{X_2}{R_0}\right)^2} \right] \tag{106}$$

The insertion losses obtained from equation (105) for certain specific values of X_1 are of interest at this time, namely, when $X_1 = 0$, $X_1 = \infty$ and $|X_1| = R_0 \frac{K-1}{\sqrt{K}}$. Successively inserting these values into equation (105) we have

$$\left[\text{I.L.}\right]_{X_1 = 0} = 10 \log 1 = \text{zero} \tag{107}$$

$$\left[\text{I.L.}\right]_{X_1 = \infty} = 10 \log K^2 = 20 \log K \tag{108}$$

$$\left[\text{I.L.}\right]_{X_1 = R_0 \frac{K-1}{\sqrt{K}}} = 10 \log K \tag{109}$$

In other words, the insertion loss of equalizers employing these inverse arms varies over the range from zero to a maximum value equal to 20 log K. When the loss (20 log K) is obtained, the equalizer reactive elements are ineffective and the equalizer circuits reduce to the attenuator circuits shown in Figure 167. When $|X_1|$ takes on the special value of $R_0 \frac{K-1}{\sqrt{K}}$ the insertion loss, as shown by equation (109), is one-half its maximum value. In the following work the symbol f_b is used to denote the frequency for which $|X_1| = R_0 \frac{K-1}{\sqrt{K}}$, or what is the same thing, $|X_2| = R_0 \frac{\sqrt{K}}{K-1}$. This frequency f_b is employed as a design parameter for the following networks. It is referred to as the half-loss frequency. The frequency f_a previously used to indicate the three db loss point cannot be used where the inverse arms of equalizers contain some resistive elements for the reason that such equalizers need not have even this amount of loss

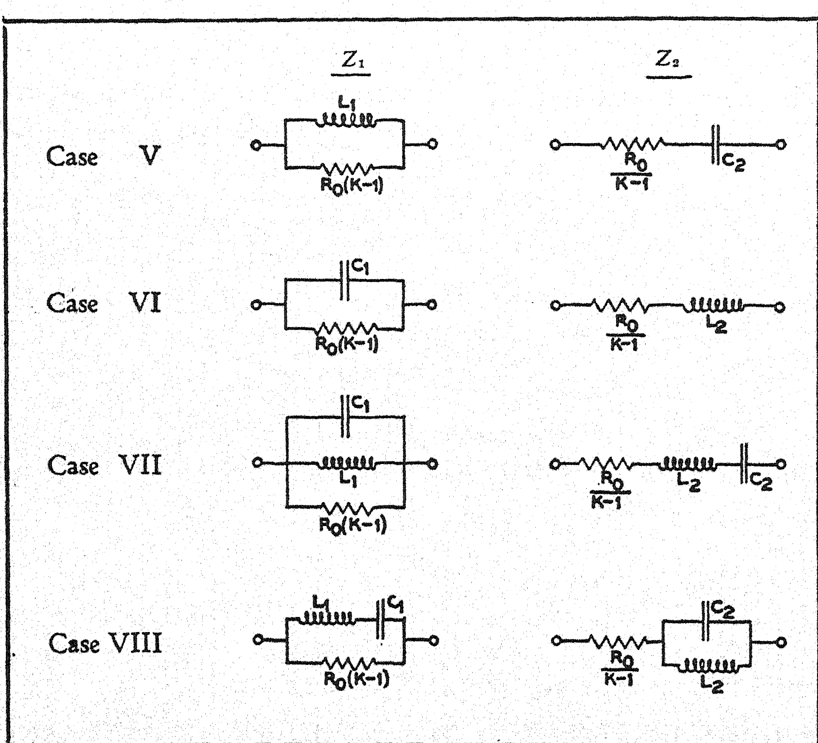
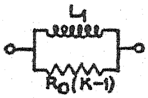
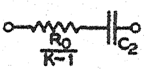


Figure 175 — Pairs of resistive and reactive inverse arms

Immediately following, insertion loss formulae are developed and chart data presented using the pairs of inverse arms of Figure 175.

(a) Case V

When $Z_1 =$ 

and $Z_2 =$ 

In this case X_2 of the Z_2 arm, is by means of equation (51),

$$X_2 = X_B \frac{f_B}{f}$$

and again letting $X_B = R_0 \frac{\sqrt{K}}{K-1}$ we have

$$X_2 = R_0 \frac{\sqrt{K}}{K-1} \frac{f_b}{f}$$

Substitution into equation (106) gives for the insertion loss of networks employing these inverse arms

$$\text{I. L.} = 10 \log \left[1 + \frac{K^2 - 1}{1 + K\left(\frac{f_b}{f}\right)^2} \right] \tag{110}$$

In the same manner as before, C_2 and L_1 become

$$\frac{1}{2\,\pi f_b C_2} = R_0 \frac{\sqrt{K}}{K-1}$$

$$C_2 = \frac{1}{2\,\pi f_b R_0} \frac{K-1}{\sqrt{K}} = \frac{1}{\omega_b R_0} \frac{K-1}{\sqrt{K}}$$

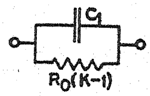
(111)

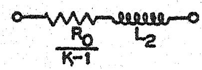
$$L_1 = R_0^2 C_2 = \frac{R_0}{\omega_b} \frac{K-1}{\sqrt{K}}$$

Chart VI shows the insertion loss formula of (110) plotted for various values of the pad loss; that is, for various values of K. Column

V of Chart I gives the equalizer sections obtained when these arms are used with the equalizer types of Figure 166.

(b) Case VI

When $Z_1 =$ 

and $Z_2 =$ 

For this pair of inverse arms we have by means of equation (48)

$$Z_2 = \frac{R_0}{K-1} + jX_s\frac{f}{f_s} \tag{112}$$

At the frequency f_b where the insertion loss of equalizers employing these arms is $10 \log K$, the value of X_s is $R_0\frac{\sqrt{K}}{K-1}$. The reactive component X_2 of equation (112) then becomes

$$X_2 = \frac{R_0\sqrt{K}}{K-1}\,\frac{f}{f_b}$$

Substitution of this value of X_2 into equation (106) gives for the insertion loss formula

$$\text{I.L.} = 10 \log\left[1 + \frac{K^2-1}{1+K\left(\frac{f}{f_b}\right)^2}\right] \tag{113}$$

This insertion loss formula is now in convenient form for plotting. Chart VII shows such a family of curves plotted against the ratio f/f_b for different values of K. At zero frequency, where $f/f_b = 0$, the formula reduces to the pad loss value of $20 \log K$. Chart VII is arranged so that the pad loss varies in one db steps from two db to fourteen db, inclusive. Each curve of the charts is an insertion loss characteristic for a given value of K and plotted against the ratio f/f_b, where f_b is the frequency where the insertion loss is one-half the pad loss. The values of L_2 and C_1 in terms of R_0, K and f_b are secured as follows: At the

frequency f_b the reactance X_2, as already stated, is $R_0 \dfrac{\sqrt{K}}{K-1}$.

Then we have

$$2\,\pi\, f_b\, L_2 = R_0 \frac{\sqrt{K}}{K-1}$$

or

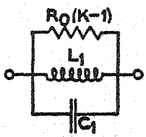
$$L_2 = \frac{R_0}{2\,\pi\, f_b}\,\frac{\sqrt{K}}{K-1} = \frac{R_0}{\omega_b}\,\frac{\sqrt{K}}{K-1} \tag{114}$$

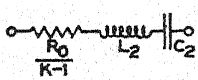
and from the inverse relationship of L_2 and C_1, we have $L_2/C_1 = R_0^2$ or

$$C_1 = \frac{L_2}{R_0^2} = \frac{1}{R_0\,\omega_b}\,\frac{\sqrt{K}}{K-1} \tag{115}$$

Column VI of Chart I shows the equalizers obtained when this pair of inverse arms is used with the equalizer types of Figure 166.

(c) Case VII

When $Z_1 =$ 

and $Z_2 =$ 

The reactive component X_2 of the above Z_2 network is by means of equation (55),

$$X_2 = X_S \frac{\dfrac{f}{f_R} - \dfrac{f_R}{f}}{s - \dfrac{1}{s}}$$

Where f_R is the frequency of resonance of L_2 and C_2, X_S is the reactance at the frequency f_S and $s = \dfrac{f_R}{f_S}$. Letting $X_S = R_0 \dfrac{\sqrt{K}}{K-1}$ at the frequency f_b and letting $s = b = f_R/f_b$ we have

$$X_2 = R_0 \frac{\sqrt{K}}{K-1}\,\frac{\dfrac{f}{f_R} - \dfrac{f_R}{f}}{b - \dfrac{1}{b}}$$

By means of equation (106) we obtain the following insertion loss formula for networks using these inverse arms:

$$\text{I. L.} = 10 \log \left[1 + \frac{K^2 - 1}{1 + K \left(\dfrac{\dfrac{f}{f_R} - \dfrac{f_R}{f}}{b - \dfrac{1}{b}} \right)^2} \right] \tag{116}$$

The values of L_2 and C_2 are secured from equations (56) and (57) if we remember that we have set up by definition the relations $X_S = R_0 \dfrac{\sqrt{K}}{K-1}$, $a = b$ and $f_S = f_b$. We have

$$L_2 = \frac{R_0}{\omega_b} \frac{\sqrt{K}}{K-1} \frac{1}{b^2 - 1} \qquad C_2 = \frac{1}{R_0\, \omega_b} \frac{K-1}{\sqrt{K}} \frac{b^2-1}{b^2} \tag{117}$$

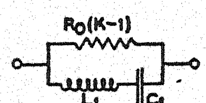
And since by the inverse relationship between Z_1 and Z_2, $L_1 = C_2 R_0^2$ and $C_1 = L_2/R_0^2$ we have

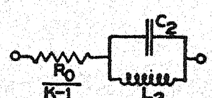
$$L_1 = \frac{R_0}{\omega_b} \frac{K-1}{\sqrt{K}} \frac{b^2-1}{b^2} \qquad C_1 = \frac{1}{\omega_b R_0} \frac{\sqrt{K}}{K-1} \frac{1}{b^2-1} \tag{118}$$

The insertion loss formula of (116) is expressed in terms of the frequency ratio f/f_b and the parameters K and b. To chart this equation it is necessary to make curves for selected values of parameters. This has been done in Charts VIII to XVII, inclusive, where K has values which make the pad loss vary from three db to fourteen db in one db steps.

Column VII of Chart I gives the equalizers obtained when these inverse arms are used with equalizer types of Figure 166.

(d) Case VIII

When $Z_1 =$ 

and $Z_2 =$ 

Development of the insertion loss formula for equalizers using this

pair of inverse arms proceeds in the same manner as for the previous cases. We have for the reactance X_1 of the impedance Z_1

$$X_1 = X_S \frac{\dfrac{f}{f_R} - \dfrac{f_R}{f}}{s - \dfrac{1}{s}}$$

At the frequency f_b, $X_S = R_0 \dfrac{K-1}{\sqrt{K}}$ and $s = b$. This gives

$$X_1 = R_0 \frac{K-1}{\sqrt{K}} \frac{\dfrac{f}{f_R} - \dfrac{f_R}{f}}{b - \dfrac{1}{b}}$$

When this value of X_1 is used with equation (105) the insertion loss formula becomes

$$\text{I.L.} = 10 \log \left[1 + \frac{K^2 - 1}{1 + K\left(\dfrac{b - \dfrac{1}{b}}{\dfrac{f}{f_R} - \dfrac{f_R}{f}}\right)^2} \right] \qquad (119)$$

By means of formulae (56) and (57), and remembering that

$$\frac{L_2}{C_1} = \frac{L_1}{C_2} = R_0^2$$

we have for the electrical constants of the inverse arms

$$L_1 = \frac{R_0}{\omega_b} \frac{K-1}{\sqrt{K}} \frac{1}{b^2 - 1} \qquad L_2 = \frac{R_0}{\omega_b} \frac{\sqrt{K}}{K-1} \frac{b^2 - 1}{b^2}$$

$$C_1 = \frac{1}{\omega_b R_0} \frac{\sqrt{K}}{K-1} \frac{b^2 - 1}{b^2} \qquad C_2 = \frac{1}{\omega_b R_0} \frac{K-1}{\sqrt{K}} \frac{1}{b^2 - 1}$$

Charts XVIII to XXVII, inclusive, give insertion loss characteristics for equalizers using these inverse arms as computed from formula (119). As in the previous case, each set of curves is for a particular pad loss; the range covered varying from three db to fourteen db in one db steps. Column VIII of Chart I shows the equalizer designs obtained when these inverse arms are used with the fundamental equalizer types.

CHART I

(a)

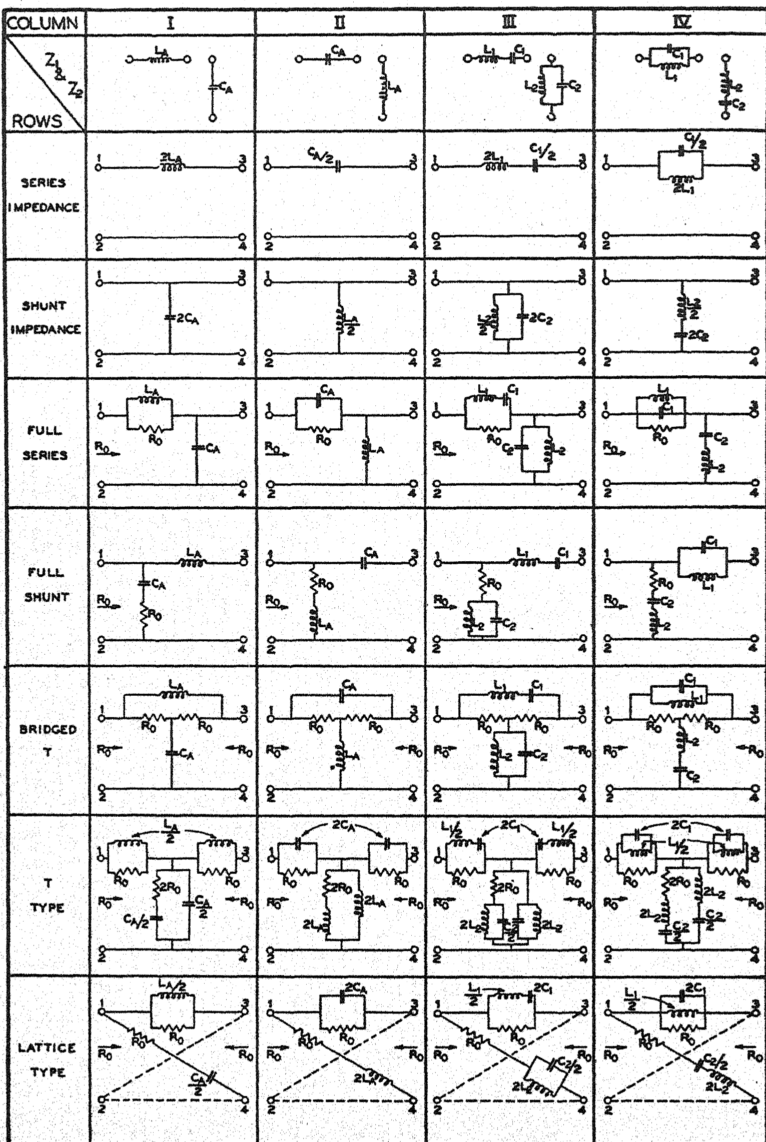

COLUMN
I
II
III
IV
Z_1 & Z_2
ROWS
SERIES IMPEDANCE
SHUNT IMPEDANCE
FULL SERIES
FULL SHUNT
BRIDGED T
T TYPE
LATTICE TYPE

CHART I

(b)

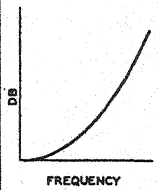
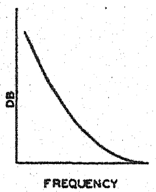
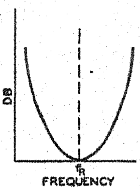
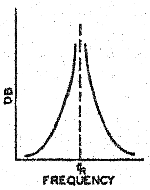
COLUMN	I	II	III	IV
INSERTION LOSS CHARACTERISTIC				
REFER TO	CHART II	CHART III	CHART IV	CHART V
CURRENT RATIO $\left\lvert\left[I_L/I_L'\right]^2\right\rvert$	$1+\left[\frac{f}{f_a}\right]^2$	$1+\left[\frac{f_a}{f}\right]^2$	$1+\left[\frac{\frac{f}{f_R}-\frac{f_R}{f}}{a-\frac{1}{a}}\right]^2$	$1+\left[\frac{a-\frac{1}{a}}{\frac{f}{f_R}-\frac{f_R}{f}}\right]^2$
DESIGN FORMULAE	$L_A=\frac{R_0}{2\pi f_a}=\frac{R_0}{\omega_a}$ $C_A=\frac{1}{2\pi f_a R_0}=\frac{1}{\omega_a R_0}$ $f_a=\frac{1}{2\pi\sqrt{L_A C_A}}$ $R_0=\sqrt{\frac{L_A}{C_A}}$		$L_1=L_A\frac{1}{a^2-1}$ $L_2=L_A\frac{a^2-1}{a^2}$ $C_1=C_A\frac{a^2-1}{a^2}$ $C_2=C_A\frac{1}{a^2-1}$	$L_1=L_A\frac{a^2-1}{a^2}$ $L_2=L_A\frac{1}{a^2-1}$ $C_1=C_A\frac{1}{a^2-1}$ $C_2=C_A\frac{a^2-1}{a^2}$
NOTES	f_R=RESONANT FREQUENCY OF Z_1 & Z_2 ARMS f_a=FREQUENCY OF 3 DB INSERTION LOSS f=ANY FREQUENCY $a=\frac{f_R}{f_a}$=DEFINED AS GREATER THAN UNITY R_0=EQUALIZER RESISTANCE INSERTION LOSS=10 LOG $\left\lvert\left[\frac{I_L}{I_L'}\right]^2\right\rvert$ L=INDUCTANCE IN HENRIES C=CAPACITANCE IN FARADS			

CHART I

(c)

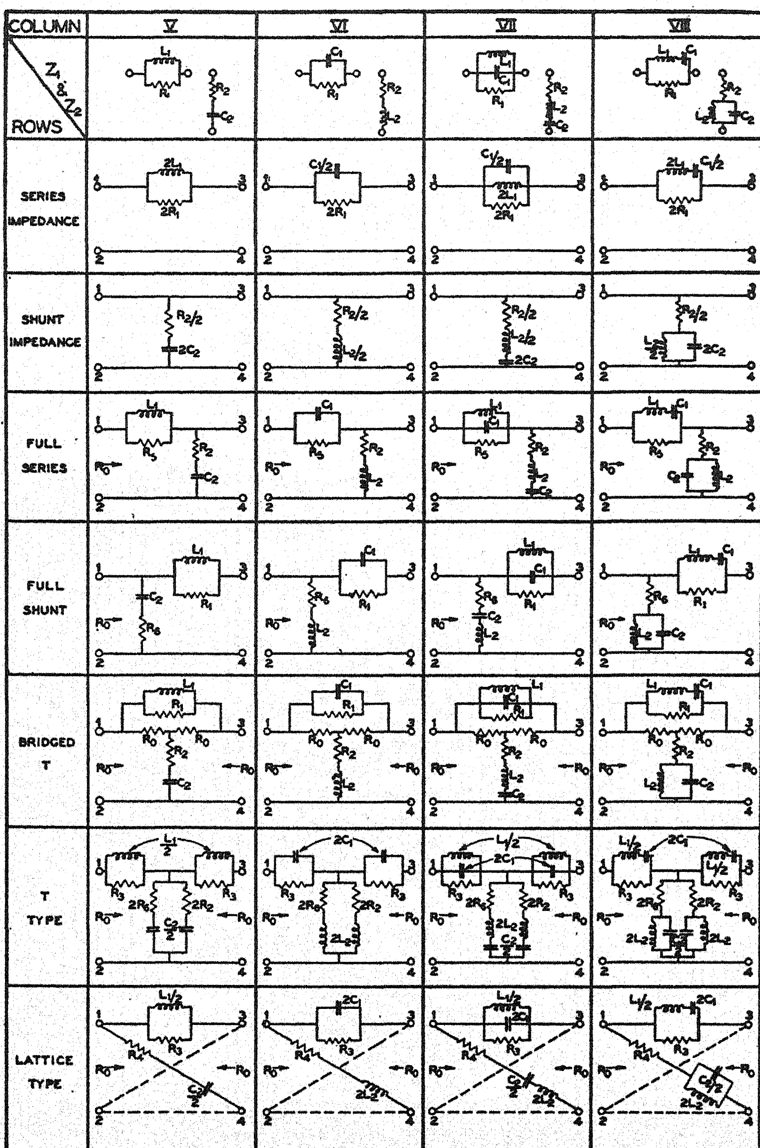

COLUMN
V
VI
VII
VIII
Z1 & Z2
ROWS
SERIES IMPEDANCE
SHUNT IMPEDANCE
FULL SERIES
FULL SHUNT
BRIDGED T
T TYPE
LATTICE TYPE

CHART I

(d)

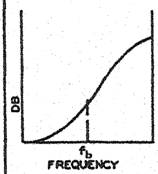
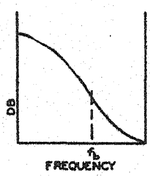
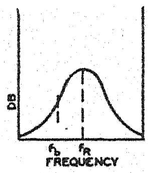
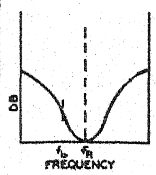
COLUMN	V	VI	VII	VIII
INSERTION LOSS CHARACTERISTIC				
REFER TO	CHART VI	CHART VII	CHARTS VIII TO XVII	CHARTS XVIII TO XXVII
CURRENT RATIO $\left\lvert\left[I_L/I'_L\right]^2\right\rvert$	$1+\dfrac{K^2-1}{1+K\left(\dfrac{f_b}{f}\right)^2}$	$1+\dfrac{K^2-1}{1+K\left(\dfrac{f}{f_b}\right)^2}$	$1+\dfrac{K^2-1}{1+K\left[\dfrac{\frac{f}{f_R}-\frac{f_R}{f}}{b-\frac{1}{b}}\right]^2}$	$1+\dfrac{K^2-1}{1+K\left[\dfrac{b-\frac{1}{b}}{\frac{f}{f_R}-\frac{f_R}{f}}\right]^2}$
DESIGN FORMULAE	$L_1=L_B\dfrac{K-1}{\sqrt{K}}$ $L_2=L_B\dfrac{\sqrt{K}}{K-1}$ $C_1=C_B\dfrac{\sqrt{K}}{K-1}$ $C_2=C_B\dfrac{K-1}{\sqrt{K}}$		$L_1=L_B\dfrac{K-1}{\sqrt{K}}\,\dfrac{b^2-1}{b^2}$ $L_2=L_B\dfrac{\sqrt{K}}{K-1}\,\dfrac{1}{b^2-1}$ $C_1=C_B\dfrac{\sqrt{K}}{K-1}\,\dfrac{1}{b^2-1}$ $C_2=C_B\dfrac{K-1}{\sqrt{K}}\,\dfrac{b^2-1}{b^2}$	$L_1=L_B\dfrac{K-1}{\sqrt{K}}\,\dfrac{1}{b^2-1}$ $L_2=L_B\dfrac{\sqrt{K}}{K-1}\,\dfrac{b^2-1}{b^2}$ $C_1=C_B\dfrac{\sqrt{K}}{K-1}\,\dfrac{b^2-1}{b^2}$ $C_2=C_B\dfrac{K-1}{\sqrt{K}}\,\dfrac{1}{b^2-1}$
	FOR ALL NETWORKS $R_0=\sqrt{\dfrac{L_B}{C_B}}$ $\quad R_1=R_0(K-1)$ $\quad R_2=R_0\dfrac{1}{K-1}$ $\quad R_6=R_0\dfrac{K}{K-1}$ $R_3=R_0\dfrac{K-1}{K+1}$ $\quad R_4=R_0\dfrac{K+1}{K-1}$ $\quad R_5=R_0\dfrac{K-1}{K}$ $L_B=\dfrac{R_0}{2\pi f_b}=\dfrac{R_0}{\omega_b}$ $\quad C_B=\dfrac{1}{2\pi f_b R_0}=\dfrac{1}{\omega_b R_0}$ $\quad f_b=\dfrac{1}{2\pi\sqrt{L_B C_B}}$			
NOTES	f_R=RESONANT FREQUENCY OF Z_1 & Z_2 ARMS f_b=FREQUENCY OF ONE-HALF PAD LOSS f=ANY FREQUENCY $b=\dfrac{f_R}{f_b}$=DEFINED AS GREATER THAN UNITY INSERTION LOSS=10 LOG $\left\lvert\left[\dfrac{I_L}{I'_L}\right]^2\right\rvert$ PAD LOSS=MAXIMUM LOSS=20 LOG K L=INDUCTANCE IN HENRIES C=CAPACITANCE IN FARADS R_0=EQUALIZER RESISTANCE			

CHART II

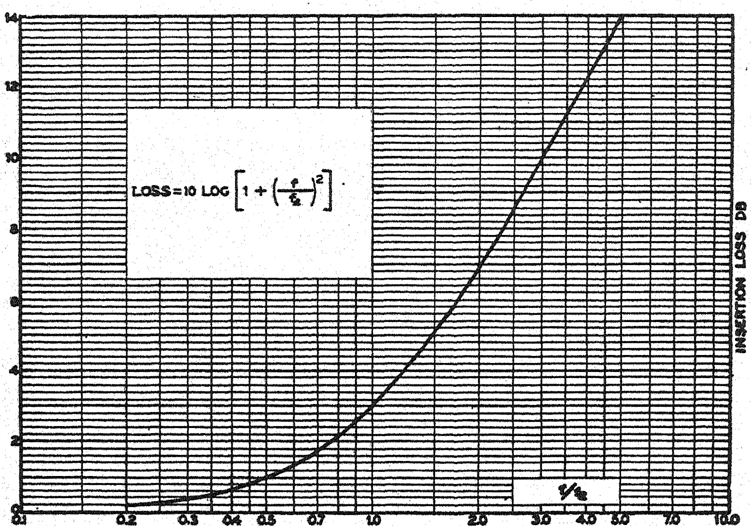


CHART III

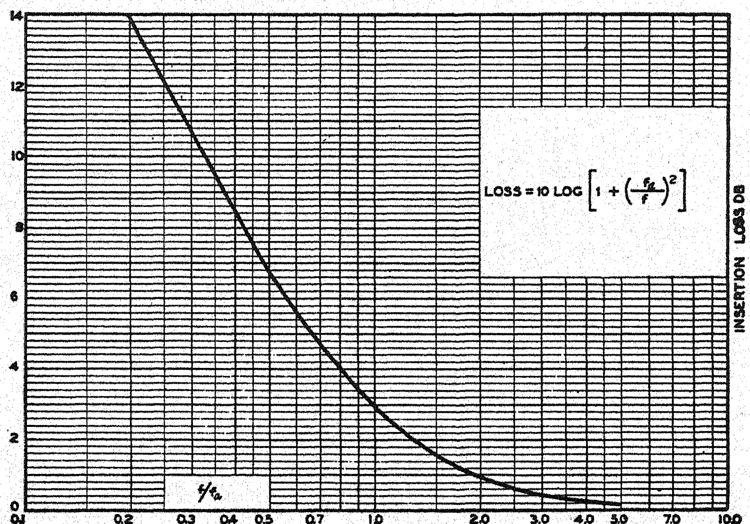
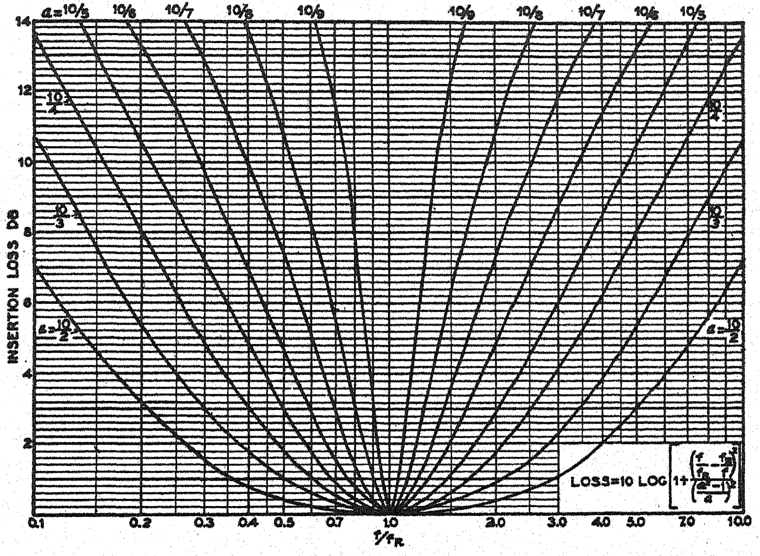


CHART IV



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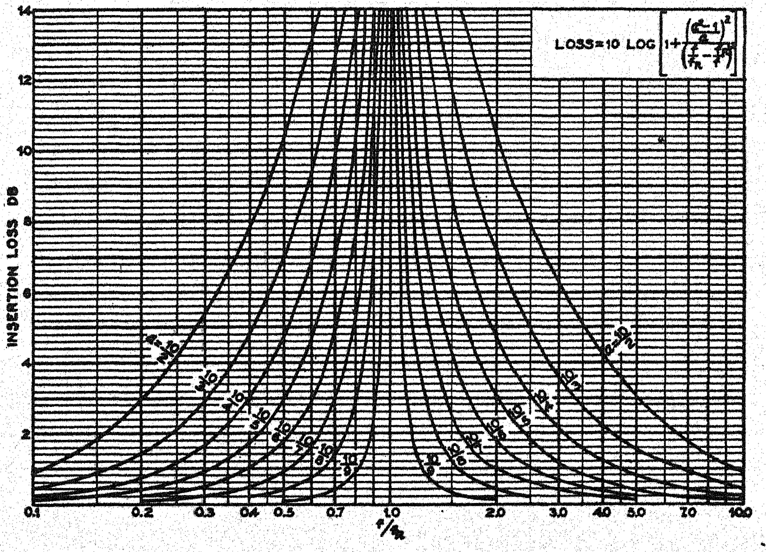
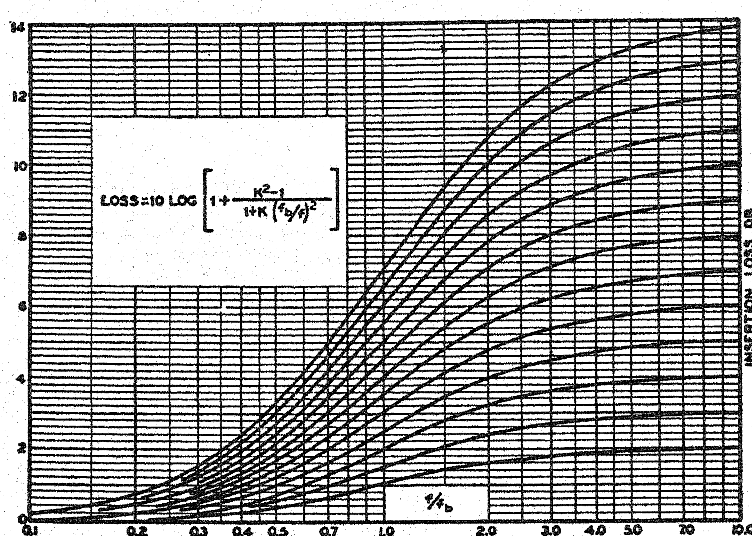
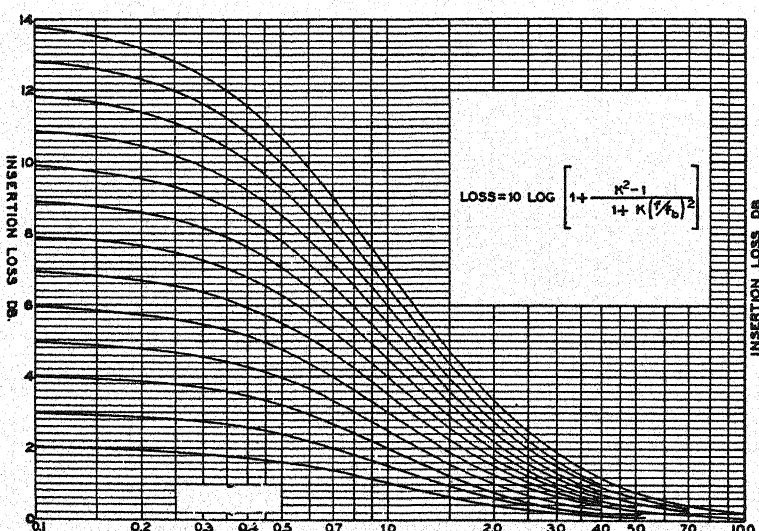


CHART VI



f/f_b

CHART VII



f/f_b

CHART VIII

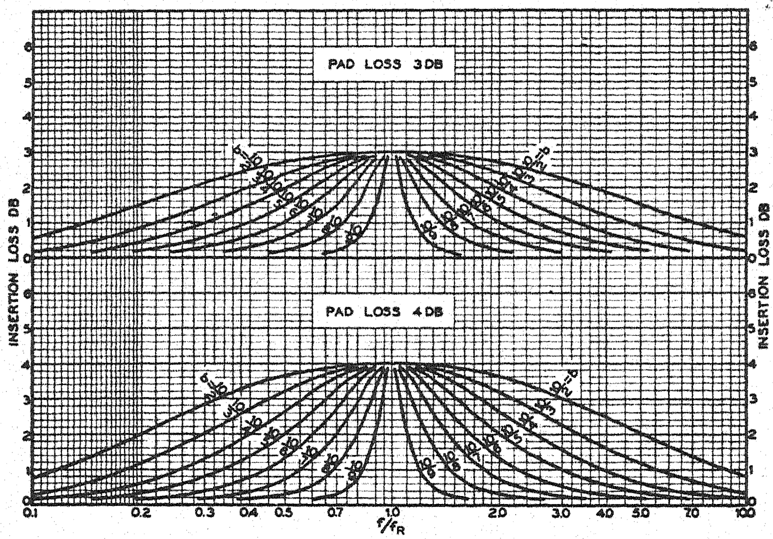


CHART IX

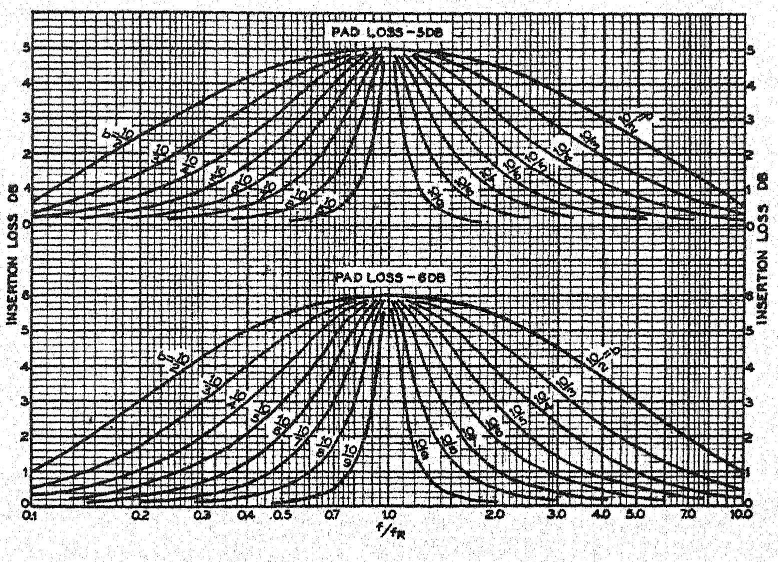


CHART X

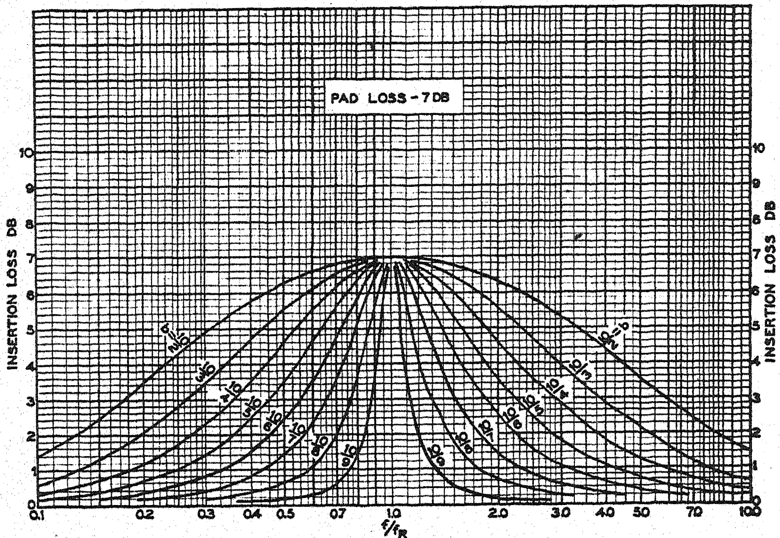
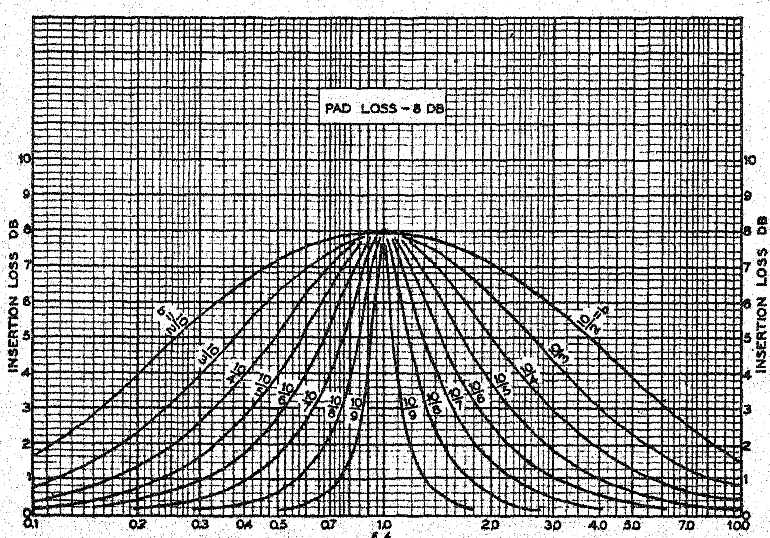


CHART XI



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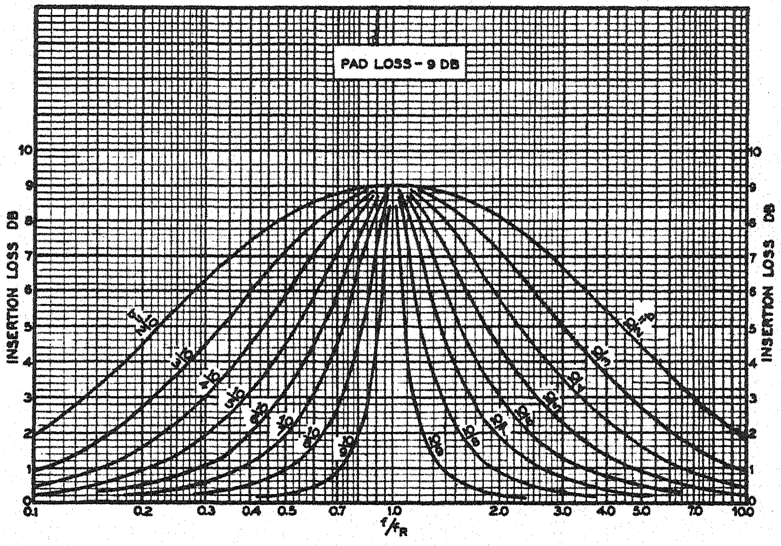

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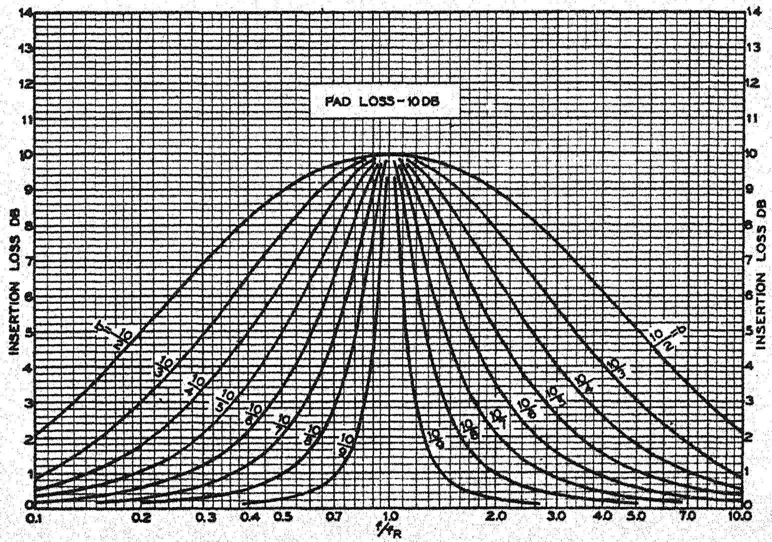


CHART XIV

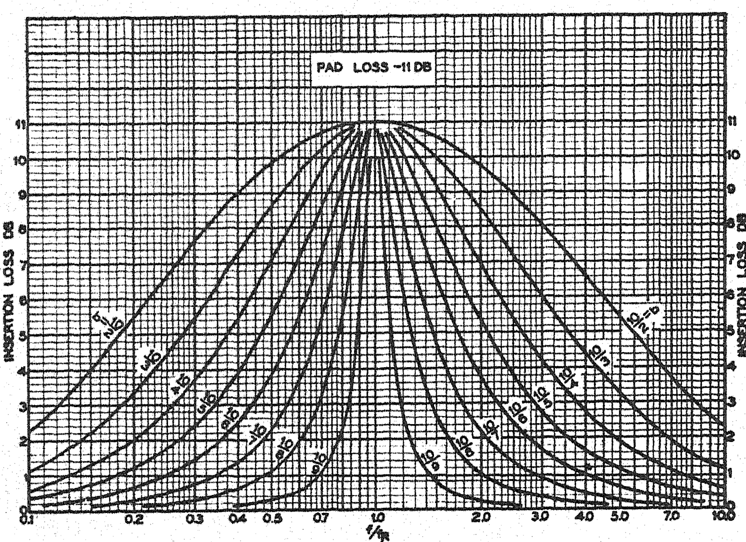


CHART XV

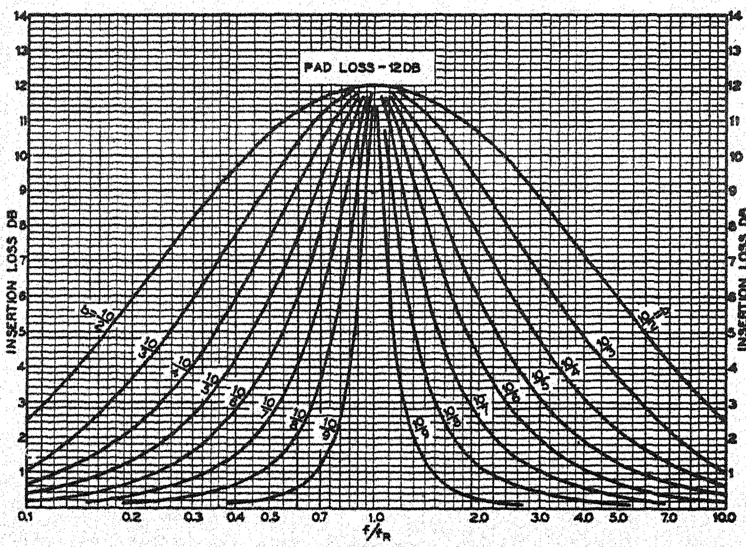


CHART XVI

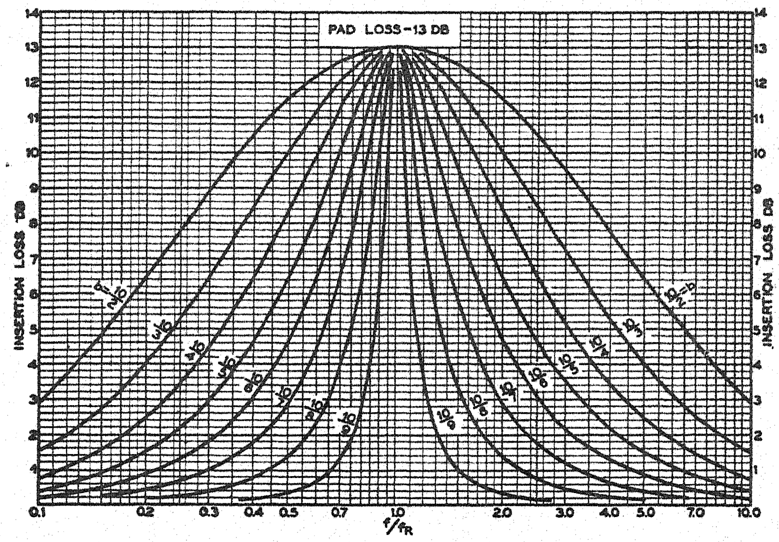
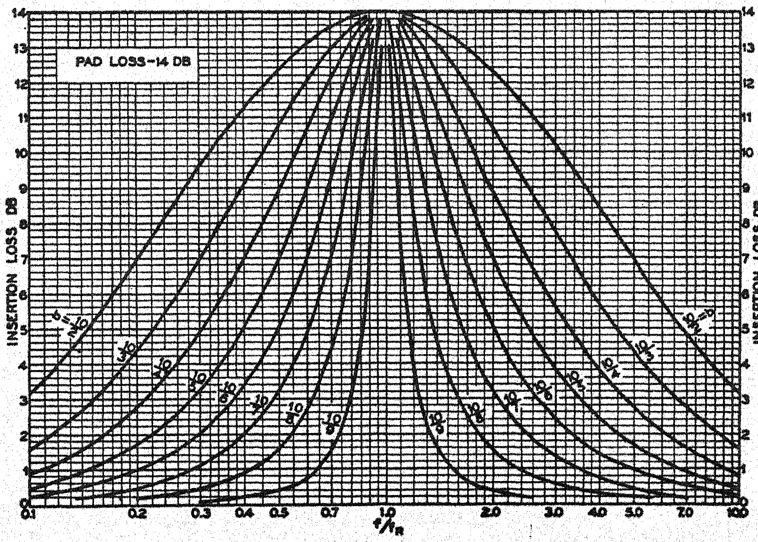


CHART XVII



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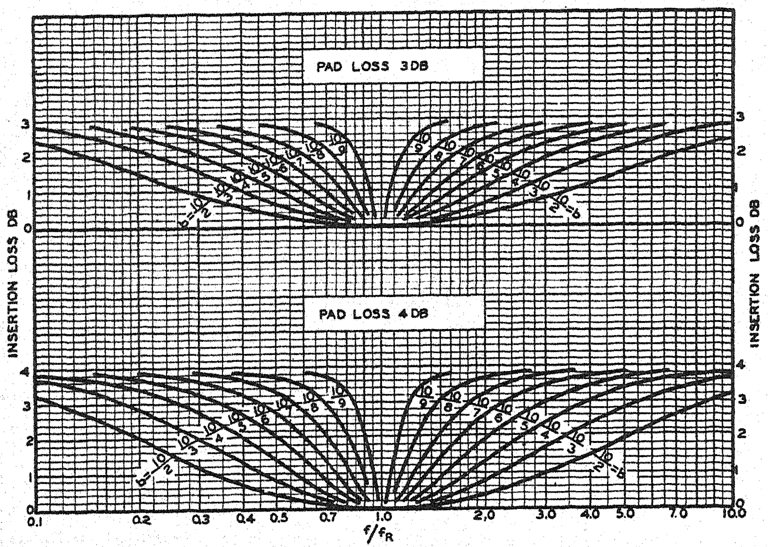

CHART XIX